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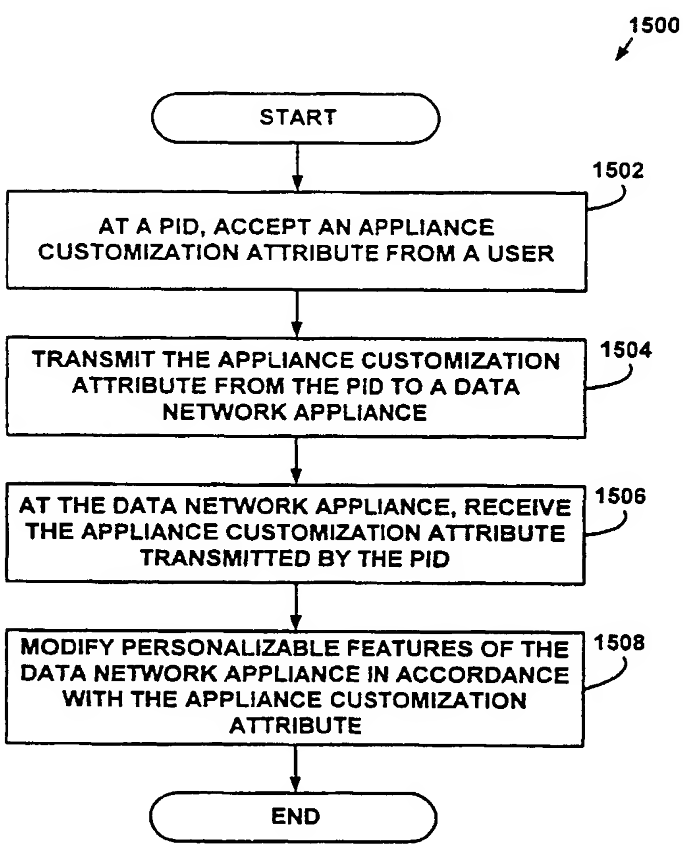

(57) Abstract: Systems and methods for personalizing a data network appliance on a data network telephony system are provided. An appliance customization attribute is provided by a user to a data network appliance, such as a data network telephone. The appliance customization attribute indicates the user's desired configuration for the data network appliance. For example, the user may specify a particular audio clip, such as one that states the user's name, to be played when the data network appliance receives an incoming call for the user. Similarly, the data network appliance may be customized to play a music clip when the user is placed on hold during a call. In a preferred embodiment, the appliance customization attribute is stored on a portable information device, such as a PDA (Personal Digital Assistant). When the user wishes to become registered to a particular data network appliance, the user may cause the portable information device to transmit the appliance customization attribute to the data network appliance to personalize the data network appliance.

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Personalizing a Data Network Appliance on a Data Network Telephony System

BACKGROUND OF THE INVENTION

A. Field of the Invention

The present invention is related to providing customizable communication services on a network. In particular, the present invention relates to personalizing a data network appliance in a data network telephony system.

B. Description of the Related Art

For many years, telephone service providers on the Public Switched Telephone Network (PSTN) provided their customers nothing more than a telephone line to use to communicate with other subscribers. Over time, telephone service providers have enhanced their service by providing Custom Local Area Signaling Service (CLASS) features to their customers. Similar communication services are provided by a Private Branch Exchange (PBX), which is typically implemented in a nonresidential setting.

The CLASS features permit customer subscribers of the features to tailor their telephone service according to individual needs. Some of the more popular CLASS features are:

- *Call blocking:* The customer may specify one or more numbers from which he or she does not want to receive calls. A blocked caller will hear a rejection message, while the callee will not receive any indication of the call.
- *Call return:* Returns a call to the most recent caller. If the most recent caller is busy, the returned call may be queued until it can be completed.
- *Call trace:* Allows a customer to trigger a trace of the number of the most recent caller.
- *Caller ID:* The caller's number is automatically displayed during the silence period after the first ring. This feature requires the customer's line to be equipped with a device to read and display the out-of-band signal containing the number.
- *Caller ID blocking:* Allows a caller to block the display of their number in a callee's caller ID device.

• *Priority ringing:* Allows a customer to specify a list of numbers for which, when the customer is called by one of the numbers, the customer will hear a distinctive ring.

• *Call forwarding:* A customer may cause incoming calls to be automatically forwarded to another number for a period of time.

A customer subscriber to a CLASS feature may typically activate and/or de-activate a CLASS feature using "*" directives (e.g., *69 to automatically return a call to the most recent caller). CLASS features may also be implemented with the use of out-of-band data. CLASS feature data is typically transmitted between local Class-5 switches using the Signaling System #7 (SS7).

Local Exchange Carriers (LECs) and other similar organizations maintain CLASS offices that typically contain a database entry for each customer. The database allows specification of the CLASS features a customer has subscribed to, as well as information, such as lists of phone numbers, associated with those features. In some cases, customers may edit these lists on-line via a touch-tone interface. A list of all phone numbers that have originated or terminated a call with each customer is often included in the CLASS office database. For each customer, usually only the most recent number on this list is stored by the local Class-5 switch.

A Private Branch Exchange (PBX), is a stored program switch similar to a Class-5 switch. It is usually used within a medium-to-large-sized business for employee telephony service. Since a PBX is typically operated by a single private organization, there exists a wide variety of PBX services and features. Custom configurations are common, such as integration with intercom and voice mail systems. PBX's typically support their own versions of the CLASS features, as well as other features in addition to those of CLASS. Most PBX features are designed to facilitate business and group communications.

A summary of typical PBX features includes:

• *Call transfer:* An established call may be transferred from one number to another number on the same PBX.

• *Call forwarding:* In addition to CLASS call forwarding, a PBX number can be programmed to automatically transfer a call to another number when the first number does not answer or is busy.

- *Camp-on queuing:* Similar to PSTN call return, a call to a busy number can be queued until the callee can accept it. The caller can hang up their phone and the PBX will ring them when the callee answers.
- *Conference calling:* Two or more parties can be connected to one another by dialing into a conference bridge number.
- *Call parking:* An established call at one number can be put on hold and then reestablished from another number. This is useful when call transfer is not warranted.
- *Executive override:* A privileged individual can break into an established call. After a warning tone to the two participants, the call becomes a three-way call.

While the CLASS and PBX features have enhanced the offerings of service providers that use the PSTN, the features are nevertheless limited in their flexibility and scope. The effect to the user is that the features become clumsy and difficult to use. For example, in order to use the Call Forwarding function, the user must perform the steps at the user's own phone prior to moving to the location of the telephone to which calls will be forwarded. A more desirable approach, from the standpoint of usefulness to the user, would be to perform the steps at the telephone or other device to which calls will be forwarded.

Much of the lack of flexibility of the PSTN features is due to the inflexible nature of the PSTN system itself. One problem with the PSTN is that the terminal devices (*e.g.* telephones) lack intelligence and operate as "dumb" terminals on a network having the intelligence in central offices. Most PSTN telephones are limited in functional capability to converting the analog signals they receive to sound, converting the sound from the handset to analog signals, generating the appropriate dial tones when a key on the keypad is pressed, and ringing when there is an incoming call.

Some PSTN telephones have a display device and a display function to display specific information communicated from intelligent agents in the PSTN network using the PSTN signaling architecture. For example, some PSTN telephones have a display function to enable the Caller ID feature. Even such PSTN telephones are limited however by the closed PSTN signaling architecture, which prohibits access by the PSTN telephones to the network signaling protocols. A PSTN telephone having a display function is effectively limited to displaying text, again, as a "dumb" terminal.

The Internet presents a possible solution for distributing intelligence to telephony terminal devices. In Internet telephony, digitized voice is treated as data and transmitted across a digital data network between a telephone call's participants. One form of Internet telephony uses a telephony gateway/terminal where IP telephony calls are terminated on the network. PSTN telephones are connected by a subscriber line to the gateway/terminal at the local exchange, or at the nearest central office. This form of Internet telephony provides substantial cost savings for users. Because the PSTN portion used in Internet telephony calls is limited to the local lines on each end of the call, long distance calls may be made for essentially the cost of a local call. Notwithstanding the costs savings provided by this form of Internet telephony, it is not much more flexible than the PSTN with respect to providing enhancements and features to the basic telephone service.

In another form of Internet telephony, telephones are connected to access networks that access the Internet using a router. The telephones in this form of Internet telephony may be substantially more intelligent than typical PSTN telephones. For example, such a telephone may include substantially the computer resources of a typical personal computer.

Data network telephones and the data network (*e.g.* Internet) system in which they operate, however, lack a substantial infrastructure and service providers for providing telephone service.

It would be desirable to incorporate CLASS and PBX features into a data network telephony system that uses a data network such as the Internet.

It would be desirable to provide new features and enhancements to telephony service that accommodate and conform to users' needs.

It would also be desirable to provide features and capabilities to telephone service that create new opportunities for users and for service providers.

It would also be an advantage for a data network telephony system to provide user customization of a data network telephony system.

It would also be advantageous to allow a user to utilize information stored in a user's Portable Information Device (PID), e.g. a Personal Digital Assistant (PDA), to personalize a data network appliance.

SUMMARY OF THE INVENTION

The present invention is directed toward personalizing a data network appliance to conform to a user's specifications. Preferably, the appliance customization attribute is transmitted by a portable information device associated with the user to the data network appliance. Alternatively, the user may directly enter an appliance customization attribute into the data network appliance.

In a first embodiment of the present invention, a system for personalizing a data network appliance in a data network telephony system is provided. The system includes a data network appliance connected to a data network, and a portable information device associated with a user. The data network appliance is operable to communicate a voice signal as voice-over-data packets on a voice-over-data channel over the data network. Similarly, the data network appliance is also operable to convert voice-over-data packets communicated on the voice-over-data channels to voice signals to be played back to the user. The portable information device stores an appliance customization attribute, such as a user identifier, a personal ring tone to be played by the data network appliance when an incoming call is received for the user, or on-hold music, for example. The portable information device may transmit the appliance customization attribute to the data network appliance to enable the data network appliance to modify personalizable features according to the appliance customization attribute. If the appliance customization attribute is a user identifier, then, upon receiving the transmitting appliance customization attribute from the portable information device, the data network appliance may access a database located on the data network to determine further information regarding the user's appliance personalization specifications.

In another embodiment of the present invention, a method for customizing a data network appliance in a data network telephony system is provided. The method includes accepting an appliance customization attribute from a user at a data network appliance, and modifying personalizable features of the data network appliance in accordance with the appliance customization attribute. Here again, if the appliance customization attribute includes a user attribute, then the data network appliance may access a database (either local or on a data network) to identify personalizable features to modify in accordance with the user attribute. The user may directly enter the appliance customization attribute into the data network appliance, or the user may

utilize a portable information device, such as a smartcard, to transmit the appliance customization attribute to the data network appliance.

In yet another embodiment of the present invention, a method for personalizing a data network appliance on a data network telephony system is provided. The method includes receiving an appliance customization attribute at the data network appliance, and determining whether an incoming call is for the user. If the data network appliance determines that the incoming call is for the user, then the data network appliance plays a personalized ring corresponding to the appliance customization attribute received by the data network appliance. The appliance customization attribute preferably includes a personalized ring specification, and is preferably transmitted by a portable information device associated with the user.

In still yet another embodiment of the present invention, a method for personalizing a data network appliance from a portable information device is provided. The portable information device is associated with a user, and accepts an appliance customization attribute from the user. The portable information device then transmits the appliance customization attribute to a data network appliance to be personalized.

BRIEF DESCRIPTION OF THE DRAWINGS

Presently preferred embodiments of the invention are described below in conjunction with the appended drawing figures, wherein like reference numerals refer to like elements in the various figures, and wherein:

FIG. 1 is block diagram of a data network telephony system according to one embodiment of the present invention;

FIG. 2 is a block diagram showing a system for providing data network appliance personalization functions to a user on a telephony network according to an exemplary embodiment of the present invention;

FIG. 3 is a block diagram showing a system for providing data network appliance personalization functions to a user on a telephony network according to a preferred embodiment of the present invention;

FIG. 4 is a block diagram showing a system for providing data network appliance personalization functions to a user on a telephony network according to an exemplary embodiment of the present invention;

FIG. 5A is a message flow diagram showing an exemplary SIP registration operation;

FIG. 5B is a message flow diagram showing an exemplary SIP call setup operation;

FIG. 6 is a block diagram of a data network telephone according to an exemplary embodiment of the present invention;

FIG. 7 is a block diagram of a portable information device (PID) according to an exemplary embodiment of the present invention;

FIG. 8 is a stack layer diagram showing the layers of an IrDA stack;

FIG. 9 is a block diagram of a portable information device (PID) according to an alternative embodiment of the present invention;

FIG. 10 is a block and stack layer diagram illustrating the protocol stacks in an exemplary embodiment of a PID linked to a data network telephone;

FIG. 11 is block and stack layer diagram illustrating an embodiment of the present invention in which a SIP call may be established;

FIG. 12 is a flow diagram illustrating a method for personalizing a data network appliance, according to an embodiment of the present invention;

FIG. 13 is a flow diagram illustrating a method for personalizing a data network appliance, according to a preferred embodiment of the present invention;

FIG. 14 is a flow diagram illustrating a method for personalizing a data network appliance from the viewpoint of a user's PID, according to a preferred embodiment of the present invention; and

FIG. 15 is a flow diagram illustrating a method for personalizing a data network appliance, according to a preferred embodiment of the present invention.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENT

A. Related Applications

The following references to patent applications are incorporated by reference herein:

* "System And Method For Providing Telephone Service Having Private Branch Exchange Features In A Data Network Telephony System" to Schuster et al., Attorney Docket No. 99,366, filed concurrently herewith;
* "System And Method For Providing A Wireless Data Network Telephone System" to Schuster et al., Attorney Docket No. 99,590, filed concurrently herewith;
* "System And Method For Accessing A Network Server Using A Portable Information Devices Through A Network Based Telecommunication System" to Schuster et al., Attorney Docket No. 99,592, filed concurrently herewith;
* "System And Method For Accessing Radio Programs Using A Data Network Telephone In A Network Based Telecommunication System" to Schuster et al., Attorney Docket No. 99,742, filed concurrently herewith;
* "System And Method For Providing Local Information In A Data Network Telephony System" to Schuster et al., Attorney Docket No. 99,838, filed concurrently herewith;
* "System And Method For Enabling A Portable Information Device For Use In A Data Network Telephone System" to Schuster et al., Attorney Docket No. 99,741, filed concurrently herewith;
* "Dialing Token For Initiating A Telephone Connection In A Data Network Telephone System" to Schuster et al., Attorney Docket No. 99,375, filed concurrently herewith;
* "Flexible Dial Plan for a Data Network Telephony System" to Schuster, et al., Attorney Docket No. 99,374, filed concurrently herewith;
* "Personalized Call Announcement on a Data Network Telephony System" to Schuster, et al., Attorney Docket No. 99,597, filed concurrently herewith;

* "Proximity-Based Registration on a Data Network Telephony System" to Schuster, et al., Attorney Docket No. 99,599, filed concurrently herewith;
* "System and Method for Providing User Mobility Services on a Telephony Network" to Schuster, et al., Serial No. 09/451,388;
* "System and Method for Providing Call-Handling Services on a Telephony Network" to Schuster, et al., Serial No. 09/470,879;
* "Method Apparatus and Communication System for Companion Information and Network Appliances" to Wang, et al., Serial No. 09/181,431;
* "System and Method for Controlling Telephone Service Using a Wireless Personal Information Device" to Schuster, et al., Serial No. 09/406,321;
* "System and Method for Advertising Using Data Network Telephone Connections" to Schuster, et al., Serial No. 09/406,320;
* "System and Method for Providing User-Configured Telephone Service in a Data Network Telephony System" to Sidhu, et al., Serial No. 09/405,283;
* "System and Method for Accessing a Network Server Using a Portable Information Device Through a Network Based Telecommunication System" to Schuster, et al., Serial No. 09/406,322;
* "System and Method for Interconnecting Portable Information Devices Through a Network Based Telecommunication System" to Schuster, et al., Serial No. 09/406,152;
* "System and Method for Enabling Encryption on a Telephony Network" to Schuster, et al., Serial No. 405,981;
* "System and Method for Associating Notes with a Portable Information Device on a Network Telephony Call" to Schuster, et al., Serial No. 09/406,151;
* "System and Method for Providing Shared Workspace Services Over a Telephony Network" to Schuster, et al., Serial No. 09/406,298;
* "System and Method for Providing Service Provider Configurations for Telephones in a Data Network Telephony System" to Schuster, et al., Serial No. 09/406,066;
* System and Method for Using a Portable Information Device to Establish a Conference Call on a Telephone Network" to Schuster, et al., Serial No. 09/406,128;

* "Multiple ISP Support for Data Over Cable Networks" to Ali Akgun, et al., Serial No. 09/321,941;
* "Method and System for Provisioning Network Addresses in a Data-Over-Cable System" to Ali Akgun, et al., Serial No. 09/218,793; and
* "Network Access Methods, Including Direct Wireless to Internet Access" to Yingchun Xu, et al., Serial No. 08/887,313.

B. PID-Enabled Data Network Telephony System

FIG. 1 is a block diagram showing an exemplary embodiment of a data network telephony system 100 according to the present invention. The system includes a data network 106. A first voice communication device 108 linked to a first access network 112 via connection 111 may communicate over the data network 106 by connecting via the first access network 112. A second voice communication device 118 is linked to a second access network 114 through connection 119 and may communicate over the data network 106 by connecting via the second access network 114.

The data network 106 in the system 100 typically includes one or more Local Area Networks (LANs) connected to one another or to a Wide-Area Network (WAN), such as an Internet Protocol (IP) network, to provide wide-scale data connectivity. The data network 106 may use Voice-Over-Packet (VOP) schemes in which voice signals are carried in data packets. The network 106 may also include a connection to the Public Switched Telephone Network (PSTN) to allow for voice connections using traditional circuit switching techniques. In one embodiment, the data network 106 may include one or more LANs such as Ethernet LANs and may support data transport protocols for performing Voice-over-Internet-Protocol (VoIP) techniques on the Internet. For further details regarding VoIP, see the information available through the website of the Internet Engineering Task Force (IETF) at www.ietf.org. In addition, an Internet Telephony gateway may be included within the system 100 to allow for voice connections to users connected by subscriber lines at a PSTN Central Office. Other data besides voice data, such as video data, may also be communicated over the data network 106.

The voice communication devices 108 and 118 typically include a voice input, a voice output, and a voice processing system and may be data network telephones (described further below with reference to FIG. 6), which are telephones adapted for use with a data network. The voice processing system converts voice sound to digital data signals that are communicated on a voice connection over the data network. The voice processing system also converts digital data signals received from the voice connection to voice sound. The voice communication devices 108 and 118 typically include a central processing unit and memory to store and process computer

programs. Additionally, each voice communication device typically includes a unique device address or network address, such as an IP address, in memory to uniquely identify it to the data network 106 and to permit data packets to be routed to the device. In one embodiment, the voice communication devices 208 and 118 are data network appliances. A data network appliance is a general purpose device connected to a data network. Through software, either resident on the data network appliance or downloaded via the data network, a data network appliance may be customized for specific applications, such as for voice communications. An Internet appliance is an example of a data network appliance.

A PID 110 is shown linked to the first voice communication device 108 via link 109, and may enable customizable communications over the data network 106 via the first access network 112. The PID 110 includes user attributes stored in a user information database. The user attributes may contain such information as a user identifier, schedule information, information about contacts, and other information that is associated with a user of the PID 110. The PID 110 preferably includes a user interface allowing a user to easily enter and retrieve data. In a preferred embodiment, the user interface includes a pressure-sensitive display that allows a user to enter input with a stylus or other device. An example of a PID with such an interface is a PDA (Personal Digital Assistant), such as one of the Palm™ series of PDAs offered by 3Com® Corporation. Alternatively, the PID 110 may be a form of smart card, in which the user attributes are programmed into the card with the assistance of a programming device. In such a case, the user attributes might not be easily modified by the user when the user is not in the presence of the programming device. The PID 110 may include other functionality, such as wireless phone, two-way radio, digital camera, or digital audio recording functionality, for example. To assist with customizing a call announcement and/or a data network appliance, the PID 110 preferably includes at least one call announcement attribute and/or at least one appliance customization attribute, respectively.

Link 109 is a point-to-point link, and may be entirely or partially wireless, or may be a hard-wired connection. Preferably, the link 109 is a wireless link, such as an infrared link specified by the Infrared Data Association (IrDA) (see irda.org for further information) or a radio frequency (RF) link such as the Bluetooth system (see www.bluetooth.com for further information). However, the point-to-point link can

also be a hardwired connection, such as an RS-232 or Universal Serial Bus (USB) serial port connection. An example of a serial port connection is a docking cradle or a synchronizing cable connection. For the provision of proximity-based registration of a user to a voice communication device, a wireless omnidirectional link is preferred. Link management application may be present on the PID and the voice communication devices 108 and 118 to assist with link setup, maintenance, and teardown.

In one embodiment, the voice communication devices 108 and 118 each include a handset with a receiver and transmitter similar or identical to handsets of traditional circuit-switched telephones. A console on which the handset sits may include the voice processing system, a display, and a keypad, for example.

In a preferred embodiment, a portion of each of the voice communication devices 108 and 118 utilizes an NBX 100™ communication system phone offered by 3Com® Corporation. In alternative embodiments, the voice communication devices 108 and 118 may include any device having voice communications capabilities. For example, a personal computer having a microphone input and speaker output may also be used to implement the voice communication devices 108 and 118. Other configurations are also intended to be within the scope of the present invention.

The details relating to operation of the voice communication devices 108 and 118 depend on the nature of the data network 106, the access networks 112 and 114 connecting the voice communication devices 108 and 118 to each other and/or to other network entities, and the PID 110. The access networks 112 and 114 typically include any high bandwidth network adapted for data communications, i.e. a network having greater than 64,000 bits-per-second (bps) bandwidth. The access networks 112 and 114 may link to the voice communication devices 108 and 118 using an Ethernet LAN, a token ring LAN, a coaxial cable link (*e.g.* CATV adapted for digital communication), a digital subscriber line (DSL), a twisted pair or fiberoptic cable, an Asynchronous Transfer Mode (ATM) link, an Integrated Services Digital Network (ISDN) link, or a wireless link, for example. In embodiments that may not require bandwidth greater than 64,000 bps, the access networks 112 and 114 may also include the PSTN and link the voice communications devices 108 and 118 by an analog modem. Further details regarding specific implementations are described below, with reference to FIGs. 2 through 17.

C. System for Personalizing a Data Network Appliance on a Data Network Telephony System

One advantage of the network telephony system 100 is that it may be used to provide personalized call announcement functions to users. In one embodiment, the PID 110 is used to register a user to a voice communication device, such as the voice communication device 108. The user may transfer personalization attributes, such as call announcement attributes or appliance customization to the voice communication device 108 through the link 109. The voice communication device 108 then sets up the call with voice communication devices associated with the communication partners selected by the PID user associated with the voice communication device 108. Similarly, the user may be called at the data network appliance to which he is registered. The transferred personalization attributes enable the user to make or receive customized calls. For example, a personalized call announcement may be invoked to replace the standard bell "ring" that is heard on a typical POTS phone. The caller's name may be audibly announced to the called party instead, in one embodiment. As another example, if an appliance customization attribute is transmitted from the user's PID 110 to the voice communication device 108, then the voice communication device 108 may behave in accordance with the user's preferences. For example, if several users are registered to the same voice communication device 108, then the appliance customization attribute may be used by an application in the voice communication device 108 to play personalized ring tones for incoming calls, depending on the intended party-to-be-called. The called-user's name may be audibly announced in one embodiment. Other possible voice communication device personalizations may include customizing speed-dial keys to a user's preferred set, adjusting speaker volume to the user's preferred level, or adjusting a video display in a video phone implementation.

As an alternative to using the PID 110 to transfer personalization attributes, the user may directly enter any personalization attributes into the voice communication device 108. For example, this may be done as part of a programming operation, in which input buttons or a voice response system is used in combination with a display output or audio output to place the user's preferences into a memory within the voice communication device. These personalization attributes may be

stored in the memory of the voice communication device 108 or at another device on the data network 106, such as a registration server or storage archive, for example.

Once a call is set up, data can be transferred between the voice communication devices 108 and 118. PIDs, such as the PID 110, associated with the parties to the call may also be used to communicate information across the data network 106. For example, the PID 110 linked to the first voice communication device 108 may be able to accept and display PID data entered by a user through a user interface on the PID 110. The PID data can then be communicated across the link 109 to the voice communication device 108 for transport across the first access network 112, the data network 106, and the second access network 114 to the second voice communication device 118. The PID 110 can also receive PID data and other data across the link 109 for display on the PID 110. A voice-over-data channel for communicating voice-over-data can concurrently exist with this communication of PID data over a PID data channel, in a preferred embodiment.

According to one embodiment of the present invention, the link 109 is used only for discovery, registration, verification, and/or personalization. Discovery is a process by which it is determined that the PID 110 is proximate to the voice communication device 108 (or any other similar voice communication device). Therefore, when the user of the PID 110 comes within an effective transceiver range of the voice communication device 108, one or more transmitted signals will enable discovery, allowing the user to become registered to the voice communication device 108. Registration is a process whereby the user of the PID 110 becomes associated with a voice communication device, such as the voice communication device 108. A registration server (not shown) may store user/device associations in a registration database, for example. The verification refers to the process of determining whether the user of the PID 110 is still proximate to the voice communication device 108. Verification may be similar to the discovery process. In alternative embodiment, verification is omitted, and when a user moves to a different device, the user becomes registered with the new proximate device and the registration with the prior proximate device is deleted. Personalization attributes may be transferred during discovery, registration, or verification, but is preferably transferred during registration.

1. Providing a Data Network Appliance on a Local Area Network

FIG. 2 is a block diagram showing a system 200 for personalizing a data network appliance on a LAN according to one embodiment of the present invention. System 200 includes a registration server 202 having access to a registration database 204. The registration server 202 is linked to a packet-switched local area network (LAN) 206. A voice communication device 208 is also a part of the network 206. Also shown are additional voice communication devices 212, 214, 216, and 218, which may or may not be identical to each other and voice communication device 208. The voice communication devices 208, 212, 214, 216, and 218 are each preferably able to accept information from a PID 210. A user 220 is shown as having recently moved from the voice communication device 212 to the voice communication device 208. The PDA 210 is associated with the user 220. The connections shown in FIG. 2 may be entirely or partially wireless, or they may be hard-wired connections. The LAN 206, the voice communication device 208, and the PID 210 correspond respectively to the first access network 112, the voice communication device 108, and the PID 110 shown in FIG. 1.

The LAN 206 is preferably an Ethernet LAN operating according to the IEEE 802.3 specification, which is incorporated by reference herein. The voice communication devices 208, 212, 214, 216, and 218 are preferably modified Ethernet phones. An Ethernet phone is a telephone capable of communicating through an Ethernet port.

In most cases, Ethernet phones support Internet Protocol (IP), using an IP address that is either statically configured or obtained via Dynamic Host Configuration Protocol (DHCP). An exemplary Ethernet phone, such as voice communication device 208, contains two basic parts: the signaling-stack and the media-engine. While at least several different standards (e.g. SIP, H.323, MEGACO, and MGCP) and several proprietary approaches currently exist for the signaling stack, the media is almost exclusively transported via the Real Time Protocol (RTP), which itself is carried inside of User Datagram Protocol (UDP). RTP is described in H. Schulzrinne et al., "RTP: A Transport Protocol for Real-Time Applications," IETF RFC 1889, Jan. 1996, which is incorporated herein by reference. UDP is described in J. Postel, "User Datagram Protocol," IETF RFC 768, Aug. 1980, and IP is described in J. Postel, ed., "Internet Protocol," IETF RFC 791, Sept. 1981, both of which are incorporated by reference herein.

The purpose of the signaling stack in an exemplary Ethernet phone, such as the voice communication device 208, is to set up, manage, and tear down a call. During the setup phase, the location of the endpoint is discovered, communication parameters, such as the supported voice CODEC types are determined, the voice channel is established, and other parties are invited to the call if needed. Personalized call announcement attributes may also be established, such as defining an audio clip to announce the incoming call from the caller. Such an audio clip may be played after the first ring, for example. During the management phase, other parties may be invited to the call or the existing CODEC can be changed, for example. During the teardown phase, the call is terminated. The preferred call-management protocol for the present invention is Session Initiation Protocol (SIP), which is described in M. Handley et al., "SIP: Session Initiation Protocol," IETF RFC 2543, Mar. 1999, incorporated by reference herein. Alternative call-management protocols, such as the ITU-T H.323 protocol and others, may also be used to implement the present invention.

The purpose of a media engine in an exemplary Ethernet phone is to sample the voice, encode the samples, and build the RTP packets on the sending side. On the receiver side, in addition to performing the reverse operations, the media engine also typically manages a receiver buffer to compensate for network jitter. The media engine includes the features discussed with reference to the user interface of the voice communication device 108. If video is to be supported, then the media engine performs similar operations on the video data.

The user 220 is shown as being recently relocated from voice communication device 212 to voice communication device 208. In the example illustrated by FIG. 2, voice communication device 212 may be in the user's office and voice communication device 208 may in be a conference room, for example. Prior to leaving the office, the user 220 may have been registered (associated) with the voice communication device 212 in the office, so that calls were routed to that location or were processed according to attributes associated with the user being located in the office. When the user moves to the voice communication device 208 in the conference room, it would be desirable for the user to be registered with the voice communication device 208 instead of the voice communication device 212. According to one embodiment of the present invention, the user 220 is able to register

with the voice communication device 208 by using a portable information device 210 to transmit user attributes and personalization attributes to the voice communication device 208, which may then transmit all or some of the attributes to the registration server 202. The registration database 204 may then be updated with the revised user communication-location information, and in some embodiments, personalization information.

2. Personalized a Data Network Telephone on a LAN using the Session Initiation Protocol

FIG. 3 is a block diagram showing an exemplary data network telephony system 300 according to a preferred embodiment of the present invention, in which SIP is used as the call-management protocol. Portions of the system 300 are similar to the system 200 illustrated in FIG. 2. The system 300 includes a SIP server 302 having access to a SIP database 304. The SIP server is shown with a link to a LAN 306, which is preferably an Ethernet LAN. SIP phones 308, 312, 314, 316, and 318 are Ethernet phones, and are also linked to the LAN 306. A PDA 310 serves as a PID to enable registration and personalization according to a preferred embodiment of the present invention. The number of SIP phones in the system 300 can be varied to meet the needs of the users of the system 300.

Also shown in the LAN 306 is a gateway 322 with a SIP client. The gateway 322 is preferably a VoIP gateway and is in communication with a PSTN central office 324, which provides PSTN service to a PSTN phone 326. The PSTN phone 326 is likely to be one of many PSTN phones serviced by the central office 324. Additional portions of a PSTN network have been omitted from FIG. 3 for increased clarity. The PSTN network is known by those having skill in the art of telecommunications. The gateway 322, the central office 324, and the PSTN 326 are optional and need not be included within the system 300.

A router 328 may also be connected to the LAN 306. The router 328 connects the LAN 306 to a data network 330, such as a public internet. The data network preferably includes connections to additional SIP-based clients, such as an additional SIP phone 332 and a personal computer 334 operating as a SIP client. SIP will be described in more detail with reference to FIGs. 3-7, 10, 13, and 15. The router 328,

the data network 330, and the SIP-based clients 332 and 334 are optional and need not be included within the system 300.

3. Local Area Network as an Exemplary Access Network

FIG. 4 is a block diagram showing one example of the data network telephony system 100 of FIG. 1 according to the present invention. The system 400 includes a local area network 412 connected to a data network 406 by a first router 413. A second local area network 414 is connected to the data network 406 by a second router 415. A cable network 416 is connected to the data network 406 by a third router 417. Those of ordinary skill in the art will appreciate that while FIG. 4 illustrates the access networks as two local area networks 412 and 414, and a cable network 416, other types of networks may be used. For example, the local area networks and the cable network may be replaced by ISDN, DSL, or any other high-speed data link.

The local area networks 412 and 414 provide data connectivity to their respective network elements. For example, the first LAN 412 provides data connectivity to at least a first data network telephone 408 and a first network telephony connection server 450. The second LAN 414 provides data connectivity to at least a second data network telephone 418 and a second network telephony connection server 438.The local area networks 412 and 414 in FIG. 4 are, for example, Ethernet LANs operating according to the IEEE 802.3 specification, which is incorporated by reference herein; however, other types of local area networks may also be used. The first local area network 412 uses the router 413 to provide the first data network telephone 408 and the first network telephony connection server 450 with access to the data network 406. For example, the router 413 may perform routing functions using protocol stacks that include the Internet Protocol and other protocols for communicating on the Internet. Similarly, the second local area network 414 uses the router 415 to provide the second data network telephone 418 and the second network telephony connection server 438 with access to the data network 406.

The first, second, and third network telephony connection servers 450, 438, and 437 provide telephony registration, location, and session initiation services for voice connections in which at least one of their members is a party. For example, a user of the first data network telephone 408 may register for telephony service with an

administrator of the first network telephony connection server 450 and receive a user identifier and a telephone device identifier. The user identifier and telephone device identifier may be sequences of unique alphanumeric elements that callers use to direct voice connections to the user. The network telephony connection servers register users by storing user records in registration databases (not shown in FIG. 4) associated with each of the network telephony connection servers, in response to receiving registration requests.

The call setup process and the user and telephone device identifiers preferably conform to requirements defined in a call-management protocol. The call-management protocol is used to permit a caller on the data network to connect to a user identified by a user identifier in a data network telephone call. A data network telephone call includes a call setup process and a voice exchange process. The call setup process includes steps and message exchanges that a caller and callee perform to establish the telephone call. The actual exchange of voice signals is performed by a voice data communications channel. The voice data communications channel incorporates other data transport and data formatting protocols, and preferably includes well-known data communications channels typically established over the Internet.

The call management protocol used in the system 400 is preferably the Session Initiation Protocol (SIP), which is described in M. Handley et al., "SIP: Session Initiation Protocol," IETF RFC 2543, Mar. 1999, incorporated by reference herein; however, any other such protocol may be used. Other protocols include H.323, MEGACO, the Media Gateway Control Protocol (MGCP), etc.

The network telephony connection servers 450, 438, and 437 may be used to provide telephony service for mobile users. For example, a user may be registered to use the first network telephone 408 (which is identified by its telephone identifier), but the user may move to a location near a second network telephone (not shown) on the first local area network 412. The user may re-register as the user of the second network telephone. The user would then become associated with the second network telephone. Calls that identify the user by the user's user identifier may then reach the user at the second network telephone. Alternatively, the user may move to a different access network.

4. Cable Network as an Exemplary Access Network

The system 400 in FIG. 4 also shows the cable network 416 connected to the data network 406 by a router 417. The cable network 416 provides data network access to its network elements, which in FIG. 4 include the third data network telephone 428 and the third network telephony connection server 437. A user of the third data network telephone 418 connected to the cable network 416 may communicate by telephone over the data network 406 with the users of the first and second data network telephones 408 and 418 connected to the first and second local area networks 412 and 414.

The cable network 416 may include any digital cable television system that provides data connectivity. In the cable network 416, data is communicated by radio frequency in a high-frequency coaxial cable. The cable network 416 may include a head-end and/or a central termination system that permits management of the cable connections to and from the users.

5. Providing Telephony Services

The third network telephony connection server 437 is preferably a SIP-based server that performs call initiation, maintenance, and teardown for the third data network telephone 428 connected to the cable network 416. The third network telephony connection server 437 may be similar or identical to the first and second network telephony connection servers 450 and 438 connected to the first and second local area networks 412 and 414.

The system 400 shown in FIG. 4 includes a data network telephony system that permits the first and second data network telephones 408 and 418 connected to the local area networks 412 and 414 to communicate through the data network 406 with the third data network telephone 428 connected to the cable network 416. The system shown in FIG. 4 preferably uses SIP in order to establish, maintain, and tear down telephone calls between users.

There are two major architectural elements to SIP: the user agent (UA) and the network server. The UA resides at the SIP end stations, (e.g. the data network telephones), and contains two parts: a user agent client (UAC), which is responsible for issuing SIP requests, and a user agent server (UAS), which responds to such

requests. There are three different network server types: a redirect server, a proxy server, and a registrar. The various network server types may be combined into a single server, such as the network telephony connection servers 450, 437, and 438. Not all server types are required to implement the various embodiments of the present invention. The communication services to be provided will determine which servers are present in the communication system. Preferred embodiments of the present invention may be carried out using proxy servers.

One example of a SIP operation involves a SIP UAC issuing a request, a SIP proxy server acting as end-user location discovery agent, and a SIP UAS accepting the call. A successful SIP invitation consists of two requests: INVITE followed by ACK. The INVITE message contains a user identifier to identify the callee, a caller user identifier to identify the caller, and a session description that informs the called party what type of media the caller can accept and where it wishes the media data to be sent. User identifiers in SIP requests are known as SIP addresses. SIP addresses are referred to as SIP Univeral Resource Indicators (SIP-URIs), which are of the form *sip:user@host.domain*. Other addressing conventions may also be used.

FIG. 5A is a message sequence diagram 500 illustrating an embodiment of a registration message sequence for registering a PID 502 to a data network appliance 504 (such as a voice communication device or data network telephone), using a registration server 506. The PID 502 transmits user attributes, including a user SIP URI, as an announcement message 508 to the data network appliance 504. Personalization attributes, such as call announcement attributes and/or appliance customization attributes may also be transmitted by the PID 502 to the data network appliance 504. The data network appliance 504 formats a REGISTER request 510 that includes the user's SIP URI in the TO field, the data network appliance's SIP URI in the FROM field, and the SIP URI of the registration server 506 in the REQUEST-URI field and sends the REGISTER request to the registration server 506. The registration server registers the user's SIP URI with the IP address of the data network appliance 504 and returns a 200 OK response 512 to the data network appliance 504. The data network appliance 504 may send a confirmation message 516 to the PID 502 to confirm the registration.

The message sequence of FIG. 5A applies to the case where the SIP URI for the registration server is known. Other approaches to registration are possible, such as

broadcasting to the registration multicast address "sip.mcast.net" (224.0.1.75), and are discussed in further detail in RFC 2543. RFC 2543 refers to a "location server," which may serve as the registration server discussed in FIG. 5A.

Once the user's SIP URI is registered with the registration server 506, subsequent calls to the user's SIP URI are resolved to the address of the data network appliance 504. Thus, if a call is placed to the user's SIP URI, the data network appliance 504 will "ring," alerting the user of an incoming call. If appliance customization attributes were transmitted by the PID 502 to the data network appliance 504, then the ring may be personalized, such as by audibly announcing the user's name.

By synchronizing the user's SIP URI on the PID 502 with the data network appliance 504, the user of the PID 502 becomes registered to the data network appliance 504. This allows for true portability of the user's SIP URI, since calls are forwarded to the data network appliance where the user of the PID 502 has most recently registered. In this aspect of the present invention, the PID 502 becomes an authentication token representing the user and calls are forwarded to the correct data network appliance.

It is important to note that in many cases, a caller does not want to be connected to a particular phone, but rather to a particular person. SIP URIs may be assigned to a user and stored in the user's PID device. By synchronizing the PID with a data network appliance, the user's SIP URI is registered to the local data network appliance and the SIP network. Since every incoming SIP request preferably goes through the registration server, calls are directed to the data network appliance where the user is currently located. This permits the called party to be mobile, but still be locatable by the network.

If the user of the PID 502 moves and registers with another data network appliance, the old registration may be erased and a new registration entry, referring to the new data network appliance location, is created at the registration server. This may be useful not only in a company, but in a hotel or even a home, as well. Note that more than one user can be registered with a data network appliance by synchronizing each user's PID with the data network appliance. This allows multiple users to share the same phone during a meeting, for example. When a user is finished with a meeting, the user may sign off from the phone in the meeting room, in which

case the registration server may forward all calls to a predetermined location, such as to a voice mail address or to a default data network appliance, such as a phone at the user's desk.

FIG. 5B is a message flow diagram showing an exemplary SIP call setup operation 520. A SIP caller UAC 522 sends an INVITE message 524 to a SIP callee UAS 526. (The proxy server is not shown in this illustration). The INVITE message 524 contains session description information (UAC SDP) for the caller UAC 522. The callee UAS 526 sends a 200 OK message 528 to the caller UAC 522. The 200 OK message 528 contains session description information (UAS SDP) for the callee UAS 526. The caller UAC 522 sends an ACK message 530 to the callee UAS 526 to complete the session initiation operation.

Redirect servers may be used to process an INVITE message by sending back the SIP-URI where the callee is reachable. Proxy servers perform application layer routing of the SIP requests and responses. A proxy server can either be stateful or stateless. A stateful proxy holds information about the call during the entire time the call is up, while a stateless proxy processes a message without saving information contained in the message. Furthermore, proxies can either be forking or non-forking. A forking proxy can, for example, ring several phones at once until somebody takes the call. Registrar servers are used to record the SIP address (called a SIP URI) and the associated IP address. The most common use of a registrar server is for the UAC to notify the registrar where a particular SIP URI can be reached for a specified amount of time. When an INVITE request arrives for the SIP URI used in a REGISTER message, the proxy or redirect server forwards the request correctly.

At the first local area network 412, the central registrar/proxy server, such as the first network telephony connection server 450, is the primary destination of all SIP messages trying to establish a connection with users on the first local area network 412. Preferably, the first network telephony connection server 450 is also the only destination advertised to the SIP clients outside the first local area network 412 on behalf of all the SIP clients residing on the first local area network 412. The network telephony connection server 450 relays all SIP INVITE messages to the appropriate final destination (or another SIP proxy), based on a database lookup using a first SIP database (not shown) associated with the first network telephony connection server 450. This allows mobile clients to be registered to their current locations.

Similarly, the second network telephony connection server 438 is the primary destination of all SIP messages trying to establish a connection with SIP clients such as the data network telephone 418, connected to the second local area network 414. Preferably, the second network telephony connection server 438 is also the only destination advertised to the SIP clients outside the second local area network 414 on behalf of all the SIP clients (*e.g.* data network telephones) residing on the second local area network 414. The second network telephony connection server 438 relays all SIP INVITE messages to the appropriate final destination (or another SIP proxy), based on a database lookup using a second SIP database. The third network telephony server 437 behaves similarly to the first and second network telephony connection servers 450 and 438. The use of three servers is for illustrative purposes only, and other server configurations may also be used.

The data network telephones 408, 418, and 428 in the system 400 preferably have pre-programmed device identifiers (*e.g.* phone numbers), represented as SIP-URI's that are of the form *sip: user@domain*. An example is *sip: 1234567890@3Com.com*. After power-up, each of the data network telephones 408, 418, and 428 sends a SIP REGISTER message to the default registrar, such as the network telephony servers 450, 438, and 437. When a call arrives at one of the network telephony servers 450, 438, or 437 for any of the registered SIP URIs, the server will forward the call to the appropriate destination. If a data network telephone is moved to a new location, all calls to the associated SIP URI will still be properly routed to that device. In other words, the system in FIG. 4 provides device mobility in the sense that calls will "follow" the data network telephone according to its SIP URI. This is especially useful if the data network telephone 408, 418, or 428 is running the DHCP (Dynamic Host Configuration Protocol) so that when the location is changed, the IP address is also automatically changed.

An advantage of the system in FIG. 4 is that once the call is established between data network telephones, the data network 406 provides data connectivity for up to a plurality of data communications channels. For example, the data network telephones 408, 418, and 428 can communicate voice signals as voice-over-data packets on a voice-over-data channel. The data network telephones 408, 418, and 428 may also be able to communicate PID data as PID data packets on a PID data channel. An example of PID data is graphical drawing data that is input into a PDA with a

stylus device. Another example of PID data is a call participants' contact information, which may be passed on to the other call participants. Other data types may also be communicated. If PID data is input into the PID 410, the PID data may be communicated to and from the PID 410 across link 409 to the data network telephone 408, where the PID data may be assembled into packets and disassembled from packets as part of the process for communicating the PID data packets across the data network 406 and any access networks, such as the first Ethernet LAN 412, the second Ethernet LAN 414, and the cable network 416. For example, the PID data may be communicated to and from at least one other PID (not shown) through a network device (such as a data network telephone) located in the system 400.

6. The Data Network Telephones

The data network telephones 408, 418, and 428 are preferably telephones that include an Ethernet communications interface for connection to an Ethernet port. The exemplary data network telephones in 408, 418, and 428 support the Internet Protocol (IP), using an IP address that is either statically configured or obtained by access to a Dynamic Host Configuration Protocol (DHCP) server. In a general embodiment, the data network telephone 408, 418, and 428 are data network appliances, offering a flexible set of features and functions that may be customized through resident or downloaded software.

FIG. 6 is a block diagram showing the first data network telephone 408 connected to the local area network 412 in FIG. 4. The voice communication devices 108, 118, 208, 212, 214, 216, and 218 may be implemented using the concepts shown in FIGs. 4 and 6. The data network telephone 408 in FIG. 6 is connected to the LAN 412 by a network interface 600. The network interface 600 may, for example, be a network interface card, and may be in the form of an integrated circuit. A bus 602 may be used to connect the network interface 600 with a processor 604 and a memory 606. Also connected to the processor are user interface circuitry 608 and a PID transceiver 610. The PID transceiver may be a proximity transceiver, enabling proximity-based registration. Proximity registration is described in detail in U.S. Patent Application No. ____________________, filed concurrently herewith, by Guido M. Schuster, et al., and titled "Proximity-Based Registration on a Data Network Telephony System," which is incorporated by reference herein.

The PID transceiver 610 preferably includes hardware and software to enable the data network telephone 408 to communicate with a PID, such as the PID 410. Several alternatives exist for implementing the PID transceiver 610. The examples provided here are not meant to limit the scope of the present invention. Although the PID transceiver 610 preferably includes both transmitter and receiver functionality, it may instead by solely a transmitter. A first alternative PID transceiver includes an RS-232 serial connection and associated coupling hardware and mechanisms. For example, the PID interface 610 may be a docking cradle or a cradle for a PID, such as a PDA (Personal Digital Assistant), in which information may be transferred between the PID and the data network telephone 408. A second alternative for the PID transceiver 610 includes infrared circuitry for converting signals into infrared output and for accepting infrared input. A third alternative for the PID transceiver 610 includes radio frequency circuitry for converting signals into radio frequency output and for accepting radio frequency input. The second and third alternatives provide for wireless communications between the data network telephone 408 and a PID. These three alternatives are merely examples, and additional means for implementing the PID transceiver 610 between the data network telephone 408 and a PID may also be used. Additionally, more than one alternative transceiver may be included within the same data network telephone to provide redundancy in case of failure of an interface, and to improve flexibility.

The user interface circuitry 608 includes hardware and software components to provide user input and output resources for functions in the processor 604. For example a handset, display, and keypad may be included in the data network telephone 408, as may alternative user interface mechanisms. The user interface circuitry may include a display interface 624, a keypad interface 626, an audio output interface 628, and an audio input interface 630.

For some applications, the user interface circuitry 608 may only need to support sending or receiving, but not both. The user interface circuitry 608 preferably supports the sending and receiving of at least audio information. For example, in the case where the data network telephone 408 is a voice communication device, the user interface circuitry may include a microphone, a speaker, and analog interface circuitry. A videophone implementation might also include a camera and monitor. The data network telephone 408 is not limited to telephones or videophones –

additional user interface types, for example, such as the ones needed for computer games, (e.g. a joystick, or virtual reality headset) are also contemplated as being within the scope of the present invention.

The audio input interface 630 may receive voice signals from a microphone or other audio input device and may convert the signals to digital information. The conversion preferably conforms to the G.711 *ITU-T Standard*. Further processing of the digital signal may be performed in the audio input interface 630, such as to provide compression (*e.g.* using the ITU-T G.723.1 standard) or to provide noise reduction, although such processing may also be performed in the processor 604. Alternatively, the audio input interface 630 may communicate an analog voice signal to the processor 604 for conversion to digital information. The audio input interface 630 may be used to record a personalized call announcement message or a personalized incoming call message according to an alternative embodiment of the present invention. The recorded message may be stored in the memory 606, at a registration server, or at a network data storage archive, for example. In one embodiment, the recorded message is converted into digital audio data and is transmitted to a user's PID to enable the user to reuse the recorded message when registering with other data network telephones. In such a case, the digital audio data may make up at least part of the personalization attributes transmitted from a PID to a data network telephone. Video and other messages may be handled in a similar manner.

The audio output interface 628 receives digital information representing voice from the processor 604 and converts the information to sound. In one embodiment, the audio output interface 628 receives information in the form of G.711 although other processing such as decompression may be performed in the audio output interface 628. Alternatively, the processor 604 may convert digital information to analog voice signals and communicate the analog voice signals to the audio output interface 628. The audio output interface 628 may be used to lay a personalized announcement message from a caller placing a call to the data network telephone 408, enabling the user of the data network telephone to hear the personalized announcement message, such as the caller's name. Similarly, if the data network telephone itself has been personalized (e.g., by an appliance customization message

being transmitted by a user's PID), then an incoming call for the user may result in a ring that is personalized to the user's preferences.

The keypad interface 626 and the display interface 624 include well-known device interfaces and respective signal processing techniques. The user interface circuitry 608 may support other hardware and software interfaces, which may be used for personalization and other functions.

The processor 604 may consist of one or more smaller processing units, including, for example, a programmable digital signal processing engine. In the preferred embodiment, the processor is implemented as a single ASIC (Application Specific Integrated Circuit) to improve speed and to economize space. The processor 604 also may include an operating system, and application and communications software, firmware, or hardware for implementing the functions of the data network telephone 408. For example, the processor may include a personalization application to assist a user with personalizing the data network telephone 408. A similar personalization application preferably exists on the user's PID. For example, a client-server configuration may be used for the pair of personalization applications. Other applications may also be processed by the data network telephone 408. The operating system may be any suitable commercially available embedded or disk-based operating system, or any proprietary operating system.

The processor 604 preferably includes a media engine 634 and a signaling stack 636 to perform the primary communications and application functions of the data network telephone 408. The purpose of the signaling stack in an exemplary data network telephone 408 is to set up, manage, and tear down a call. During the setup phase, a user may use the keypad to enter a user identifier to call. The signaling stack 636 receives the user entry and formats a request message to send to the user identified by the user identifier to initiate a telephone call. When the request message is sent, the location of the user identified by the user identifier is discovered, communication parameters, such as the supported voice CODEC types are exchanged, and a voice over data channel is established. Personalized Announcement attributes may also be transmitted to the called data network telephone as part of the request message, or as a separate message. During the management phase, other parties may be invited to the call if needed. During the teardown phase, the call is terminated.

The call-management protocol used in the exemplary data network telephone 408 is the SIP protocol. In particular, the signaling stack 636 implements a User Agent Client 638 and a User Agent Server 640, in accordance with the SIP protocol. Alternative call-management protocols, such as the ITU-T H.323 protocol and others, may also be used to implement the present invention.

Once the call is set up, the media engine 634 manages the communication over a data communications channel using a network transport protocol and the network interface 600. The media engine 634 sends and receives data packets having a data payload for carrying data and an indication of the type of data is being transported. The media engine 634 in the data network telephones 408 may sample the voice signals from the audio input 630 (or receive voice samples from the audio input 630), encode the samples, and build data packets on the sending side. On the receiver side, in addition to performing the reverse operations, the media engine also preferably manages a receiver buffer to compensate for network jitter.

The media engine 634 preferably includes hardware and software components for personalization 632, performing registration functions 642, voice-over-data functions 644, display data functions 646, and keypad output functions 648. The media engine 634 processes data that is received from the first local area network 412, and data that is to be sent over the first local area network 412. The media engine 634 and the signaling stack 636 may operate as a combination, in which the signaling stack is used for operations involving a call management protocol, such as SIP.

For data that is received from the first local area network 412, the media engine 634 may determine from the type of data in the packet whether packets contain sampled voice signals or data for performing other functions. For example, packet headers or trailers may contain an indication of data type. Packets containing sampled voice signals are processed by voice over data function 644. The voice over data function 644 preferably conforms to a protocol for formatting voice signals as digital data streams. While any suitable protocol may be used, the media (voice signal) is preferably transported via the Real Time Protocol (RTP), which itself is carried inside of User Datagram Protocol (UDP). RTP is described in H. Schulzrinne et al., "RTP: A Transport Protocol for Real-Time Applications," IETF RFC 1889, Jan. 1996, which is incorporated herein by reference. UDP is described in J. Postel, "User Datagram Protocol," IETF RFC 768, Aug. 1980, and IP is described in J. Postel, ed., "Internet

Protocol," IETF RFC 791, Sept. 1981, both of which are incorporated by reference herein.

Packets containing data for use in registering the data network telephone 408 with a network telephony server (or other registration server) are processed by the registration function 642. These packets may originate from a user's PID, and may be received by the data network telephone 408 through the PID transceiver 610. By registering to the data network telephone 408, a user may establish with the network telephony service provider that calls addressed to the user's user identifier may be connected to the data network telephone 408. Registration may occur when the data network telephone 408 sends a request to register to a service provider host, which may be located at a registration server. The service provider host may respond by setting the user's user identifier to correspond to the device identifier of the data network telephone 408, and by acknowledging the request with a status message to the data network telephone 408. In one embodiment, a request to register the data network telephone 408 to a default user is automatically sent during power-up of the data network telephone 408. As a result, the user becomes associated with the data network telephone 408. According to a preferred embodiment of the present invention, when a second user comes into proximity of the data network telephone 408, the second user will also be registered to the data network telephone 408.

Other features may be added to the registration function 642, or implemented as extensions to the registration function 642. For example, the data network telephone 408 may be provisioned to provide selected network telephony services by establishing a data connection with a service provider, requesting the selected services, and receiving data that ensures that the services have been successfully provisioned. Such services may include, for example, caller identification, call forwarding, voice mail, and any other service offered by the network telephony service provider to enhance the capabilities of the data network telephone 408. One advantage of such provisioning functions is that services may be ordered for temporary use in a manner that is convenient to the user. Personalization is an example of such provisioning.

Packets containing data for display on the display device are processed by the display data function 646. The display data function 646 may be used for displaying,

for example, the name(s) and user identifier(s) of the other party(-ies) to the call, the status of the telephone call, billing information, and other information

For data that is to be sent over the data network 406, the media engine 634 formats the data as data packets in accordance with a selected protocol. The placement of data into packets may also be performed elsewhere in the data network telephone 408. The selected protocol is preferably the protocol that is supported by the data network telephone that will receive the data for the particular type of data being transported. Tunneling may also be used for transport across multiple-protocol environments.

The voice-over-data function 644 formats voice samples, preferably according to the protocol used by the receiving data network telephone. A conversion process may also be undertaken. In one preferred embodiment, the voice over data function 644 formats voice samples as RTP packets.

The second and third data network telephones 418 and 428 are preferably similar or identical to the first data network telephone 408.

7. The Portable Information Device (PID)

FIG. 7 is a block diagram showing one embodiment of the PID 410. A bus 702 may be used to interconnect a processor 704, a memory 706, data storage 708, and user interface circuitry 710. A PID transceiver 700 enables the PID 410 to communicate with a data network telephone.

The PID transceiver 700 preferably includes hardware and software to enable the PID 410 to communicate with a data network appliance, such as the data network telephone 408. Several alternatives exist for implementing the PID transceiver 700. The examples provided here are not meant to limit the scope of the present invention. Although the PID transceiver preferably includes both transmitter and receiver functionality, it may instead by solely a transmitter (or a passive device to enable data to be read by a data network appliance). A first alternative PID transceiver includes an RF-232 serial connection and associated coupling hardware and mechanisms. For example, the PID transceiver 700 may be a docking cradle or a cradle for the PID 410, such as a PDA (Personal Digital Assistant), in which information may be transferred between the PID 410 and a data network appliance. A second alternative for the PID transceiver 700 includes infrared circuitry for converting signals into infrared output

and for accepting infrared input. A third alternative for the PID transceiver 700 includes radio frequency circuitry for converting signals into radio frequency output and for accepting radio frequency input. The second and third alternatives provide for wireless communications between the PID 410 and a data network appliance. These three alternatives are merely examples, and additional means for implementing the PID transceiver 700 between the PID 410 and a data network appliance may also be used. Additionally, more than one alternative transceiver may be included to provide redundancy in case of failure of an interface, and to improve flexibility. The PID transceiver 700 enables a user to transmit personalization attributes from the PID 410 to a data network appliance to customize the data network appliance, according to a preferred embodiment of the present invention.

The user interface circuitry 710 may include hardware and software components to provide user input and output resources for functions provided by the processor 704. The user interface circuitry preferably includes a display output 726, a display input 728, an audio input 730, and an audio output 731.

The display output 726 preferably receives digital information representing graphical or other data from the processor 704 and converts the information, such as text and/or images, for display on a graphical display, such as an LCD or TFT screen.

The display input 728 may receive PID data inputs from a user of the PID 410. The PID data inputs are preferably entered by the user with a stylus on a pressure-sensitive display screen. Alternately, a keyboard may be used to accept user input. Similarly, the display output 726 preferably displays the PID data on the display screen.

The audio input 730 may be used by the user to record an audio message for personalizing a data network appliance. For example, the user may record "This call is for Bob" as a personalized message to be played when an incoming is received for the user at a data network appliance to which the user is registered. When the user makes a call with the data network appliance, the called party may then receive this recorded announcement, informing the called party of the user's identity ("Bob"). Similarly, the audio input 730 may be a "line-in" port, to which digital music samples could be received into the PID 410. These music samples could then be transmitted to the data network appliance to be played if the user is on hold during a call, for example.

The audio output 731 may be included to assist the user in recording audio messages by allowing the user to play back recorded messages. Alternatively, a data network appliance may allow the user to play back a recorded message on a speaker of the data network appliance. The audio output 731 may also be used for other audio outputs, such as audio PID data.

The processor 704 includes an operating system and application/communication software, firmware, or hardware to implement the functions of the PID 410. The operating system may be any suitable commercially available operating system, or any proprietary operating system. For example, if the PID 410 is a PDA, then the operating system may be the Palm operating system from Palm Computing, or Windows-CE from Microsoft. The operating system and software may be stored on data storage 708 (or on memory 706). A personalization application 732 is preferably included to manage personalization functions. Similarly, other applications may manage the user's schedule and contact information, or allow a user to select communication partners to be invited to a conference call initiated by the user of the PID 410. Many other applications are also possible, and further examples of applications suitable for a PID may be found at http://www.palm.com, http://www.palmcentral.com, or http://www.tucows.com. Although the processor 704 is shown connected to the data storage 708 through a bus 702, other configurations may also be used. Similarly, the memory 706 may be alternatively configured, and may be embedded within the processor 704.

The PID 700 operates in conjunction with the PID transceiver 610 in the data network telephone 408. The PID transceivers 610 and 700 preferably operate using radio frequency signals, such as by using the unlicensed ISM band at 2.4 GHz, according to the Bluetooth specification. Bluetooth is described in the Bluetooth specification 1.0 and accompanying references and erratas, which may be found on the Bluetooth home page (http//www.bluetooth.com/) or on the Bluetooth Network (http//www.bluetooth.net/). Bluetooth is a short-range wireless communications protocol. Alternative, infrared (IrDA), a physical-wire serial connection, or other linking technologies may also be utilized by the PID transceivers 610 and 700.

The PID 410 may be able to send data to and receive data from the data network telephone 408 across a point-to-point link, such as the point-to-point link 409. A user enters PID data at the display input 728. The PID data may be processed

in the user interface circuitry 710 or it may go directly to the processor 704 or the memory 706. The processor 704 may also perform such processing functions as compression. A PID data application may be used to implement the display input, the display output, and the processing functions. The proximity transceiver 700 may be used to transmit and receive data to and from the data network telephone. Alternatively, an additional transceiver may be dedicated to PID data transceiving. Personalization attributes may be transmitted as PID data.

As an example, a drawing application may be used to accept PID data input at the display input 728 from a user drawing with a stylus on a display screen (if one exists) of the PID 410. A drawing application could then display the drawing through the display output 726 to enable the user to see a visual representation of the drawing. If the user desires to share the drawing with a second user on the system 400, where the second user is using a second PID, the PID data from the drawing application can be transmitted through the proximity transceiver 700, allowing the data to be received by the data network telephone 408. An application in the data network telephone 408 receives the PID data across the point-to-point link, and the PID data is prepared for transmission across the data network 406, such as by the media engine 634 shown in FIG. 6. Preferably the PID data is converted to PID data packets and is communicated on a PID data channel across the first LAN 412 through the router 413 across the data network 406 and eventually to a network device at which the second PID is located. The second user may then view the drawing on a display screen on the second PID.

The point-to-point link 409 may be implemented as a serial bit stream between an application in the PID 410 and an application in the first data network telephone 408. For example, the link 409 could be an infrared or radio frequency link that is implemented with minimal stack interpretation. However, the link 409 between PID 410 and the first data network telephone 408 can alternatively be implemented using all or parts of a specialized protocol, such as the Infrared Data Association (IrDA) protocol stack, where data is interpreted through the stack between application-layer processes at each end of the link.

FIG. 8 is a protocol diagram illustrating the layers of the IrDA protocol stack. An IrDA stack is implemented at each of the connection endpoints of an IrDA link. The required layers of an IrDA protocol stack are the physical layer 802, the IrLAP

layer 804, the IRLMP layer 806 and the IAS layer 808. The physical layer 802 specifies optical characteristics of the link, encoding of data, and framing for various speeds. The IrLAP (Link Access Protocol) layer 804 establishes the basic reliable connection between the two ends of the link. The IrLMP (Link Management Protocol) layer 806 multiplexes services and applications on the IrLAP connection. The IAS (Information Access Service) layer 808 provides a directory or "yellow pages" of services on an IrDA device.

The IrDA protocol also specifies a number of optional protocol layers, these protocol layers being TinyTP 810, IrOBEX 812, IrCOMM 814 and IrLAN 816. TinyTP (Tiny Transport Protocol) 810 adds per-channel flow control to keep traffic over the IrDA link moving smoothly. This important function is required in many cases. IrOBEX (Infrared Object Exchange protocol) 812 provides for the easy transfer of files and other data objects between the IrDA devices at each end of the link. IrCOMM 814 is a serial and parallel port emulation that enables existing applications that use serial and parallel communications to use IrDA without change. IrLAN (Infrared Local Area Network) 816 enables walk-up infrared LAN access for laptops and other devices. The use of the optional layers depends upon the particular application in the IrDA device. The IrDA protocol stack is defined by such standards documents as “IrDA Serial Infrared Physical Layer Link Specification”, “IrDA ‘IrCOMM’: Serial and Parallel Port Emulation over IR (Wire Replacement)”, “IrDA Serial Infrared Link Access Protocol (IrLAP)”, “IrDA Infrared Link Management Protocol (IrLMP)”, and “IrDA ‘Tiny TP’: A Flow-Control Mechanism for use with IrLMP”, and related specifications published by the IrDA and available at http://www.irda.org/standards/specifications.asp and is incorporated by reference herein.

In one embodiment, the data network telephones 408, 418, and 428 merely provide a data tunnel for the data channel attendant to the infrared links, while the IrDA protocol stack is implemented at endpoint PID devices, such as PID 410. Alternatively, IrDA stacks can be implemented in the data network telephones as well. By implementing additional layers of the IrDA protocol stack, the PID applications and the base applications in the data network telephones can be simplified because the IrDA protocol layers take over certain functions. For example, the IrDA protocol stack can be implemented at each PID used in a conference call, and the IrOBEX

layer 812 can be used to transfer text and graphics object files, such as drawings or electronic business cards, end-to-end between PID devices connected via data network telephones and networks.

FIG 9 shows a second embodiment of the PID 410, according to the present invention. The PID 410 may be part of a more complex device, such as a portable phone. The PID 410 might also be a simple data storage object, such as a smart card or a computer-readable medium such as an optical disk. Included within the PID 410 are a data storage unit 900 and a data storage interface 902.

The data storage unit 900 contains a user information database. The user information database contains personalization information, such as personalized announcement message attributes and/or appliance customization attributes. Additionally, user attributes may also be included, such as personal address and schedule information, for example.

The data storage interface 902 provides access to the data stored in the data storage unit 900. The complexity of the data storage interface 902 will depend on what reading or modifying tasks are performed by an outside device, such as a voice communication device or other data network appliance, as compared with which tasks are performed by the PID 410. If the PID 410 is a simple computer disk or smartcard, the data storage interface may be primarily mechanical in nature, so that the PID 410 is in position to read or modify user information contained in the data storage unit 900. If the PID 410 is more complex, then the data storage interface may include circuitry, possibly for reading or modifying the stored information. Infrared, magnetic, or radio frequency technology may be used to implement that data storage interface 902, for example.

Other implementations of PIDs may be used besides those described with reference to FIGs. 7 and 9. The PID will preferably include a user information database stored in data storage or memory, and should include a means for allowing an outside device to read and possibly modify the user information contained in the user information database.

Many alternative embodiments are also made possible by utilizing the PID 410. For example, the PID 410 may store and download to the data network telephone 408 the preferences of the user about the phone operation, such as the ringer volume and tone. The PID 410 may also act as a smart card, providing authentication

information for making toll calls. In another embodiment, the user of the PID 410 may program the system through the PID 410 so that, depending on the time of day, and on the datebook information in the PID 410, the phone forwarding information is dynamically updated. For example, during business hours, the default location to forward calls could be set to be the user's office, and during other hours, their cellular phone or their pager. If the PID 410 has voice playback capability, it can download voice mail and play it back off-line. On a LAN, this may be implemented as a file transfer, which is typically much faster than playing audio back. This feature would be useful if the user cannot spend too much time on the phone to check their voice mail. For example, a traveler at an airport may download their 30 minutes worth of voice mail in a few minutes, just before taking their flight, and may listen to those messages during the flight.

8. Providing User Mobility Services

FIG. 10 is a functional block diagram and protocol stack diagram illustrating an embodiment of the protocol stacks in the PID 410 and the data network telephone 408 that support link 409. In the preferred embodiment, the PID transceiver 700 in the PID 410 provides at least part of the physical layer 1000 that connects via link 409 to the PID transceiver 610 implementing a physical layer 1002 in the data network telephone 408. The data link layer 1004 in PID 410 provides data link control for link 409 in transferring data to and from a PID application client 1006. Similarly, the data network telephone 408 includes a data link layer 1012 and an application server 1008 that is configured to synchronize connection and other functions with the PID application 1006 in PID 410. The applications 1006 and 1008 may include the personalization applications.

When the user of the PID 410 wishes to register to the data network telephone 408, the application client 1006 in the PID 410 sends the user's SIP URI across the link 409 to the data network telephone 408, where it is received by the application server 1008. The application server 1008 sends the SIP URI received from the PID 410 as a REGISTER message across connection 430 and the Ethernet LAN 412 through connection 443 to the network telephony connection server 450, which is a registration server. The network telephony connection server 450 may store the SIP URI and the IP address of the associated data network telephone 408 in a SIP

registration database (not shown) so that the SIP URI is listed as being resident at the IP address of the data network telephone 408. (If the network telephony connection server 450 uses a location server for registration/location tasks, the registration information might instead be stored with such a location server). SQL (Structured Query Language) is preferred for implementing and maintaining the registration database. Once the PID 410 is registered with the network telephony connection server 450, calls to the SIP URI for the user of the PID 410 will be directed to the data network telephone 408.

FIG. 11 is a functional block and protocol stack diagram illustrating an embodiment of the present invention where a SIP connection is established from the first data network phone 408 to the second data network phone 418 through network connection 430, first access network 412, data network 406, second access network 414 and network connection 419. The routers 413 and 415 and associated connections are not shown to simplify the block diagram representation. Although only two data network telephones are shown in FIG. 11, a three-party conference call would look very similar to what is shown in FIG. 11, with the addition of an additional data network telephone. The first PID 410 and a second PID 420 are also shown for exemplary purposes, but need not be included for voice communication. Inclusion of one or more PIDs may be useful where PID data is to be communicated from one PID to a second PID.

The diagram of FIG. 11 shows how PID user data can be communicated from one PID to another PID during a call in one aspect of the present invention. The PID application 1006 in PID 410 is configured to send PID data received through the user interface 1010 through link 409 to base applications 1008 in the first data network phone 408. In this embodiment, base applications 1008 are configured to define data channels for transport to the second data network telephone 418. As illustrated, the communication system supports the use of multiple data channels.

Multiple data channels in SIP may be defined through the Session Description Protocol described in RFC 2327, herein incorporated by reference. Included in a SIP INVITE request are options for the requested connection that describe the number and type of media streams. Each media stream is described by an "m=" line in the INVITE request. For example, a request for a connection that includes an audio

stream and a bidirectional video stream using H.261 might appear as shown in Table 1.

```
v=0
o=alice 2890844526 2890844526 IN IP4 host.anywhere.com
c=IN IP4 host.anywhere.com
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000
m=video 51372 RTP/AVP 31
a=rtpmap:31 H261/90000
```

TABLE 1.

If the called device includes functionality to receive the connection as described in Table 1, then the called device will respond to the INVITE request with a 200 OK response that includes the same option values. If the called device or party is unable or unwilling to receive such a connection, then it will respond with alternative option values for the connection. See RFC 2543 for further details regarding the negotiation of connection parameters in SIP. Other call management protocols allow for similar negotiation of connection parameters. Personalization attributes may be transmitted in this manner, or by other methods.

In FIG. 11, a first data channel for voice data and a second data channel for PID user data have been negotiated by the base applications 1008 in the first data network telephone 408 and the base applications 1100 in the second data network telephone 418. The base applications 1008 and 1100 transfer voice data between the AUDIO applications, such as applications including G.711 encoders, in each phone via the first data channel. The base applications 1008 in phone 408 are also configured to send the PID data received via link 409 from PID 410 to the base applications 1100 in phone 418 via the second data channel. The base applications 1100 in phone 418 may be configured to forward the PID data received via the second data channel to a second PID 420 via a second link 421. The PID application 1102 in PID 420 then outputs the PID data received from phone 418 to the user interface 1104 for output to the user of PID 420.

The PID data in FIG. 11 can take a variety of forms. For example, the PID data can be a text file containing information about the user of PID 410, such as an electronic business card. The PID data can also be drawing data generated by graphical applications in the PIDs 410 and 420 whereby a user drawing on a touchscreen of the user interface 1010 in PID 410 generates corresponding PID data that is transmitted via the second data channel to PID 420 for display on the user interface 1104 of PID 420. The media description for the media stream can be defined during connection setup to establish a connection appropriate to the type of data being transferred. These examples represent just a few of the applications for this aspect of the present invention and should not be viewed as limiting the present invention.

In one embodiment, RTP data packets for two or more types of data are exchanged between the first data network telephone 408 and the second data network telephone 418 according to one of three possible methods. In the first method, one RTP data channel (or RTP stream) on UDP carries data packets in which both data types are present in a single split packets. Each such split packet contains (1) a source port number and a destination port number in the UDP portion, and (2) a special payload sequentially including each of the data types in the RTP portion. The special payload type can be defined in the SDP described above. Other information is also contained in each packet as well. In the second method for transmitting two or more data types, a separate RTP over UDP data channel is created for each of the different data types, and the RTP header indicates which type of data is contained in each packet. For example, voice data coded as G.711 might be assigned a payload type code of 0, while PID data is assigned a payload type code of 190. In the third method for transmitting two or more data types, a single RTP/UDP data channel (RTP/UDP stream) is created that contains data packets of two or more different types. In this method, the data types are identified in a payload type field in the RTP header of each packet, enabling an underlying application to identify which data packets are voice data packets and which data packets are PID data packets, for example.

In some embodiments of the present invention, the user interface of PID may be used to perform actual dialing operations. In other embodiments, the PID may be used to detect proximity between the user and a data network telephone and to register the user to a proximate data network telephone. The user may then use the data

network telephone user interface (such as a keypad) to input the user identifier or device identifier (such as a phone number) of the party the user wishes to call. In such a case, the PID would not be necessary for communications to proceed. Additional details for implementing embodiments in which a PID is used to initiate calls may be found in Dalgic, et al., "True Number Portability and Advanced Call Screening in a SIP-Based IP Telephony System," IEEE Communications Magazine, July, 1999, pp. 96-101; and U.S. Patent Application No. 09/181,431 "Method, Apparatus and Communications System for Companion Information and Network Appliances," filed on October 30, 1998, by Inventors Peter Si-Sheng Wang and Ismail Dalgic, assigned to 3Com Corporation, both of which are incorporated by reference herein.

D. Method for Personalizing a Data Network Appliance on a Data Network Telephony System.

1. Personalizing a Data Network Appliance from the Viewpoint of the Data Network Appliance.

FIG. 12 is a flow diagram illustrating a method 1200 for personalizing a data network appliance, according to an embodiment of the present invention. In step 1202, an appliance customization attribute is accepted from a user at a data network appliance. For example, the user may directly enter the appliance customization attribute into the data network appliance using input mechanisms on the data network appliance, such as an audio input, a touch screen display input, or other inputs. Alternatively, the user may enter the appliance customization attribute into a PID, and then transfer the appliance customization attribute from the PID to a data network appliance to which the user later becomes registered. An example of an appliance customization attribute is a digital audio file to be played when an incoming call is received by the data network appliance to which the user is registered. Another example is personalized speed dial settings for the data network appliance to enable the user to quickly make calls to the user's contacts. Other appliance customization attributes may also be used. In step 1204, the data network appliance is modified in accordance with the appliance customization attribute. A personalization application in the data network appliance assists with the customization process to modify personalizable features of the data network appliance.

FIG. 13 is a flow diagram illustrating a method 1300 for personalizing a data network appliance, according to a preferred embodiment of the present invention. In step 1302, a data network appliance receives an appliance customization attribute transmitted by a user's PID. The appliance customization attribute preferably includes a personalized ring specification to cause incoming calls for the user to be identified by a ring customized to the user's specifications. For example, the user's name may be announced when an incoming call is received by the data network appliance for the user. In step 1304, a determination is made as to whether an incoming call is for the user. If the incoming call is for the user, then the data network appliance notifies the user by playing a personalized ring corresponding to the personalized ring specification in the appliance customization attribute. If more than

one user is registered to the data network appliance, then this embodiment may enable the users to determine the intended recipient of an incoming call. Although the method 1300 is described in terms of a personalized ring specification, other personalizable features of the data network appliance may also be personalized, such as by a personalized speed dial key specification, by a personalized speaker volume specification, or by a personalized display specification, for example.

2. Personalizing a Data Network Appliance From the Viewpoint of a Portable Information Device.

FIG. 14 is a flow diagram illustrating a method 1400 for personalizing a data network appliance from the viewpoint of a user's PID, according to a preferred embodiment of the present invention. In step 1402, an appliance customization attribute from a user is accepted at a PID. For example, the user may enter the appliance customization attribute through input keys on the PID, or by recording an audio message at an audio input to the PID, or by recording an audio video message using an AV input at the PID, or by using some other input method, such as a simply file transfer from a local device or a network device. Alternatively, the user may enter an appliance customization attribute into a data network appliance, and may later cause the appliance customization attribute to be transmitted from the data network appliance to the PID for use in customizing other data network appliances to which the user registers. Personalization applications in the PID and the data network appliance may be used to coordinate personalization of the data network appliance and the transfer of an appliance customization attribute. In step 1404, the PID transmits the appliance customization attribute to a data network appliance to be personalized. This may be done via a wireless link, such as a radio frequency or infrared link, or by a serial connection between the PID and the data network appliance, for example.

3. Personalizing a Data Network Appliance Using a Portable Information Device and the Data Network Appliance.

FIG. 15 is a flow diagram illustrating a method 1500 for personalizing a data network appliance, according to a preferred embodiment of the present invention. In step 1502, a user inputs an appliance customization attribute into a PID.

Alternatively, the appliance customization attribute may be stored in the memory of the PID. In step 1504, the appliance customization attribute is transmitted from the PID to a data network appliance. In step 1506, the data network appliance receives the appliance customization attribute transmitted by the PID. In step 1508, personalizable features of the data network appliance are modified in accordance with the appliance customization attribute received from the PID.

While the invention has been described in conjunction with presently preferred embodiments of the invention, persons of skill in the art will appreciate that variations may be made without departure from the scope and spirit of the invention. For example, the access networks shown in FIG. 2 may comprise any other suitable type of local area network or service infrastructure.

In addition, protocols of various types are referenced throughout. While preferred and alternative embodiments may implement selected protocols, any suitable replacement protocol not mentioned, or any function not part of a protocol used to replace a corresponding function from a protocol may be implemented without departing from the scope of the invention.

This true scope and spirit is defined by the appended claims, interpreted in light of the foregoing.

WE CLAIM:

1. A system for personalizing a data network appliance in a data network telephony system, comprising in combination:

a data network appliance operable to communicate a voice signal as voice-over-data packets on a voice-over-data channel over a data network, wherein the data network appliance is operable to convert voice-over-data packets communicated on the voice-over-data channel to voice signals; and

a portable information device associated with a user, wherein the portable information device stores an appliance customization attribute, wherein the portable information device is operable to transmit the appliance customization attribute to the data network appliance, and wherein the appliance customization attribute specifies a personalizable feature to modify in the data network appliance.

2. The method of Claim 1, wherein the data network appliance is a data network telephone operating according to the Session Initiation Protocol standard.

3. The system of Claim 1, wherein the data network appliance is a data network telephone operating according to the ITU-T H.323 standard.

4. The system of Claim 1, wherein the data network appliance is a data network telephone operating according to the MEGACO standard.

5. The system of Claim 1, wherein the data network appliance is a data network telephone operating according to the MGCP standard.

6. The system of Claim 1, wherein the portable information device is a personal digital assistant.

7. The system of Claim 1, wherein the portable information device is a smart card.

8. The system of Claim 1, wherein the portable information device is a portable phone.

9. The system of Claim 1, wherein the portable information device is a portable digital music player, and wherein the appliance customization attribute includes an audio clip.

10. The system of Claim 1, wherein the appliance customization attribute includes a user attribute, and wherein the data network appliance is operable to access a database including the user attribute to identify the personalizable feature to modify in accordance with the user attribute.

11. The system of Claim 1, wherein the personalizable feature includes a personal ring tone to be played by the data network appliance upon receiving an incoming call for the user.

12. The system of Claim 1, wherein the personalizable feature includes a recording to be played by the data network appliance upon receiving an incoming call for the user.

13. The system of Claim 1, wherein the appliance customization attribute includes a network address corresponding to a real-time content server, and wherein the data network appliance is operable to set up a real-time data channel with the real-time content server to enable real-time content to be output to the user upon the occurrence of an event.

14. The system of Claim 13, wherein the event is an incoming call for the user.

15. The system of Claim 13, wherein the event is the user being placed on hold during a call.

16. The system of Claim 13, wherein the event is an outgoing call setup operation.

17. The system of Claim 13, wherein the personalizable feature includes at least one programmable speed dial key on the data network appliance.

18. A method for customizing a data network appliance in a data network telephony system, comprising in combination:

accepting an appliance customization attribute from a user at a data network appliance; and

modifying personalizable features of the data network appliance in accordance with the appliance customization attribute.

19. The method of Claim 18, wherein the appliance customization attribute includes a user attribute, and wherein the data network appliance accesses a database including the user attribute to identify personalizable features to modify in accordance with the user attribute.

20. The method of Claim 18, wherein the user directly enters the appliance customization attribute into the data network appliance.

21. The method of Claim 18, wherein the user transmits the appliance customization attribute from a portable information device to the data network appliance.

22. The method of Claim 18, wherein the portable information device is a personal digital assistant.

23. The method of Claim 18, wherein the data network appliance is a data network telephone operating according to the Session Initiation Protocol standard.

24. The method of Claim 18, wherein the data network appliance is a data network telephone operating according to the ITU-T H.323 standard.

25. The method of Claim 18, wherein the data network appliance is a data network telephone operating according to the MEGACO standard.

26. The method of Claim 18, wherein the data network appliance is a data network telephone operating according to the MGCP standard.

27. The method of Claim 18, wherein the step of modifying personalizable features includes storing a personal ring tone to be played by the data network appliance upon receiving an incoming call for the user.

28. The method of Claim 18, wherein the step of modifying personalizable features includes storing a recording to be played by the data network appliance upon receiving an incoming call for the user.

29. The method of Claim 18, wherein the step of modifying personalizable features includes storing a network address corresponding to a real-time content server, wherein the data network appliance sets up a real-time data channel with the real-time content server to enable real-time content to be output to the user upon the occurrence of an event.

30. The method of Claim 29, wherein the event is an incoming call for the user.

31. The method of Claim 29, wherein the event is the user being placed on hold during a call.

32. The method of Claim 29, wherein the event is an outgoing call setup operation.

33. The method of Claim 18, wherein the step of modifying personalizable features includes storing speed dial settings to program at least one speed dial key on the data network appliance.

34. A method for personalizing a data network appliance on a data network telephony system, comprising in combination:

receiving an appliance customization attribute at the data network appliance, wherein the appliance customization attribute is transmitted by a portable information device associated with a user, and wherein the appliance customization attribute includes a personalized ring specification;

determining whether an incoming call is for the user; and

playing a personalized ring corresponding to the personalized ring specification upon determining that the incoming call is for the user.

35. The method of Claim 34, wherein the data network appliance is a data network telephone operating according to the Session Initiation Protocol standard.

36. The method of Claim 34, wherein the data network appliance is a data network telephone operating according to the ITU-T H.323 standard.

37. The method of Claim 34, wherein the data network appliance is a data network telephone operating according to the MEGACO standard.

38. The method of Claim 34, wherein the data network appliance is a data network telephone operating according to the MGCP standard.

39. The method of Claim 34, wherein the portable information device is a personal digital assistant.

40. The method of Claim 34, wherein the portable information device is a smart card.

41. The method of Claim 34, wherein the portable information device is a portable phone.

42. The method of Claim 34, wherein the portable information device is a portable digital music player.

43. A method for personalizing a data network appliance from a portable information device associated with a user, comprising in combination:

accepting an appliance customization attribute from the user at the portable information device; and

transmitting the appliance customization attribute from the portable information device to the data network appliance.

44. The method of Claim 43, wherein the data network appliance is a data network telephone operating according to the Session Initiation Protocol standard.

45. The method of Claim 43, wherein the data network appliance is a data network telephone operating according to the ITU-T H.323 standard.

46. The method of Claim 43, wherein the data network appliance is a data network telephone operating according to the MEGACO standard.

47. The method of Claim 43, wherein the data network appliance is a data network telephone operating according to the MGCP standard.

48. The method of Claim 43, wherein the portable information device is a personal digital assistant.

49. The method of Claim 43, wherein the portable information device is a smart card.

50. The method of Claim 43, wherein the portable information device is a portable phone.

51. The method of Claim 43, wherein the portable information device is a portable digital music player.

52. The method of Claim 43, wherein the step of accepting the appliance customization attribute includes recording an audio sample.

53. The method of Claim 43, wherein the step of accepting the appliance customization attribute includes recording a video sample.

54. The method of Claim 43, wherein the step of accepting the appliance customization attribute includes storing a network address corresponding to a content server, wherein the content server is operable to provide content.

55. The method of Claim 54, wherein the network address is a web address, wherein the content server is a web server, and wherein the content is a digital data stream.

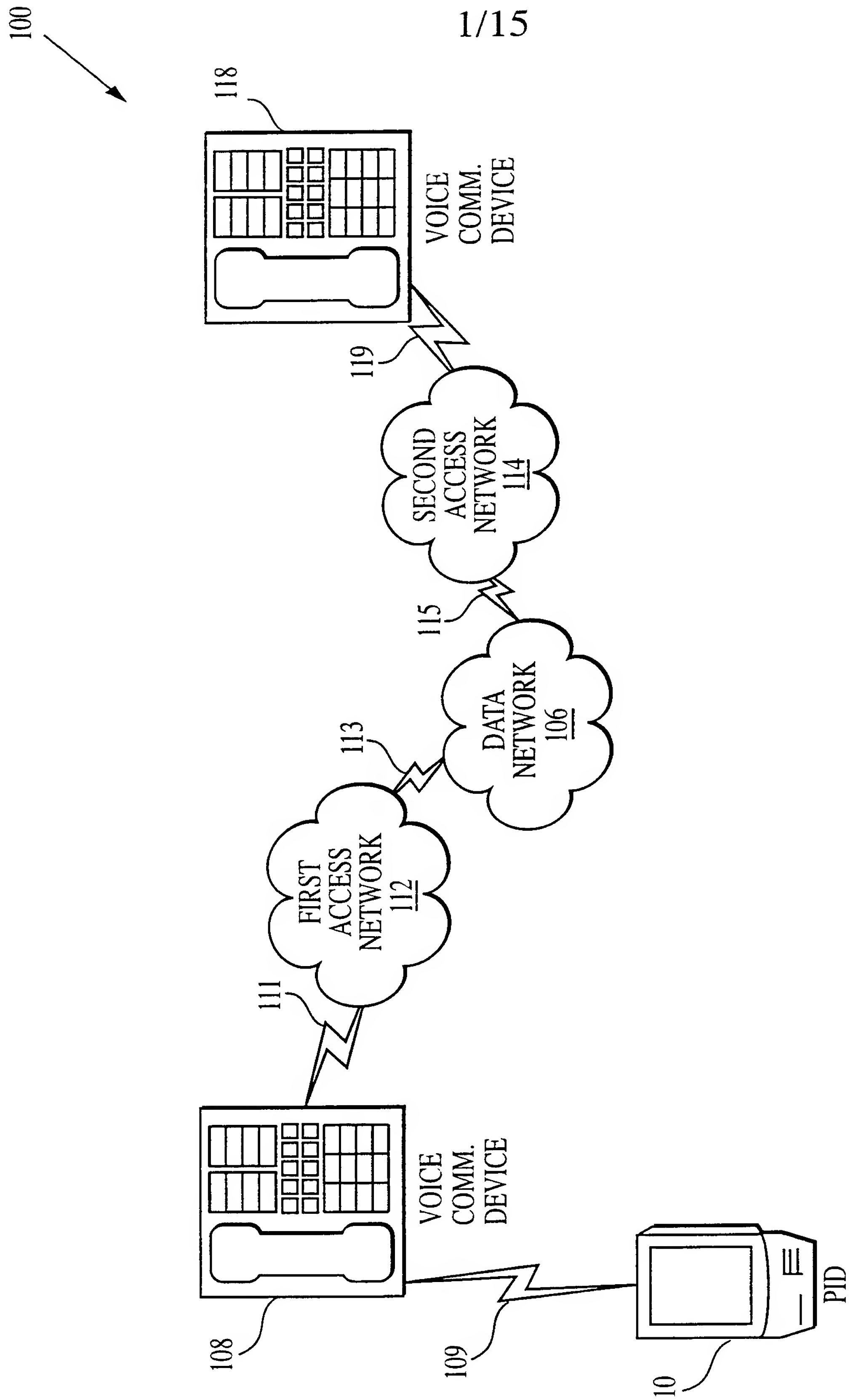
100
118
VOICE COMM. DEVICE
119
SECOND ACCESS NETWORK 114
115
DATA NETWORK 106
113
FIRST ACCESS NETWORK 112
111
VOICE COMM. DEVICE
108
109
PID
110

FIG. 1

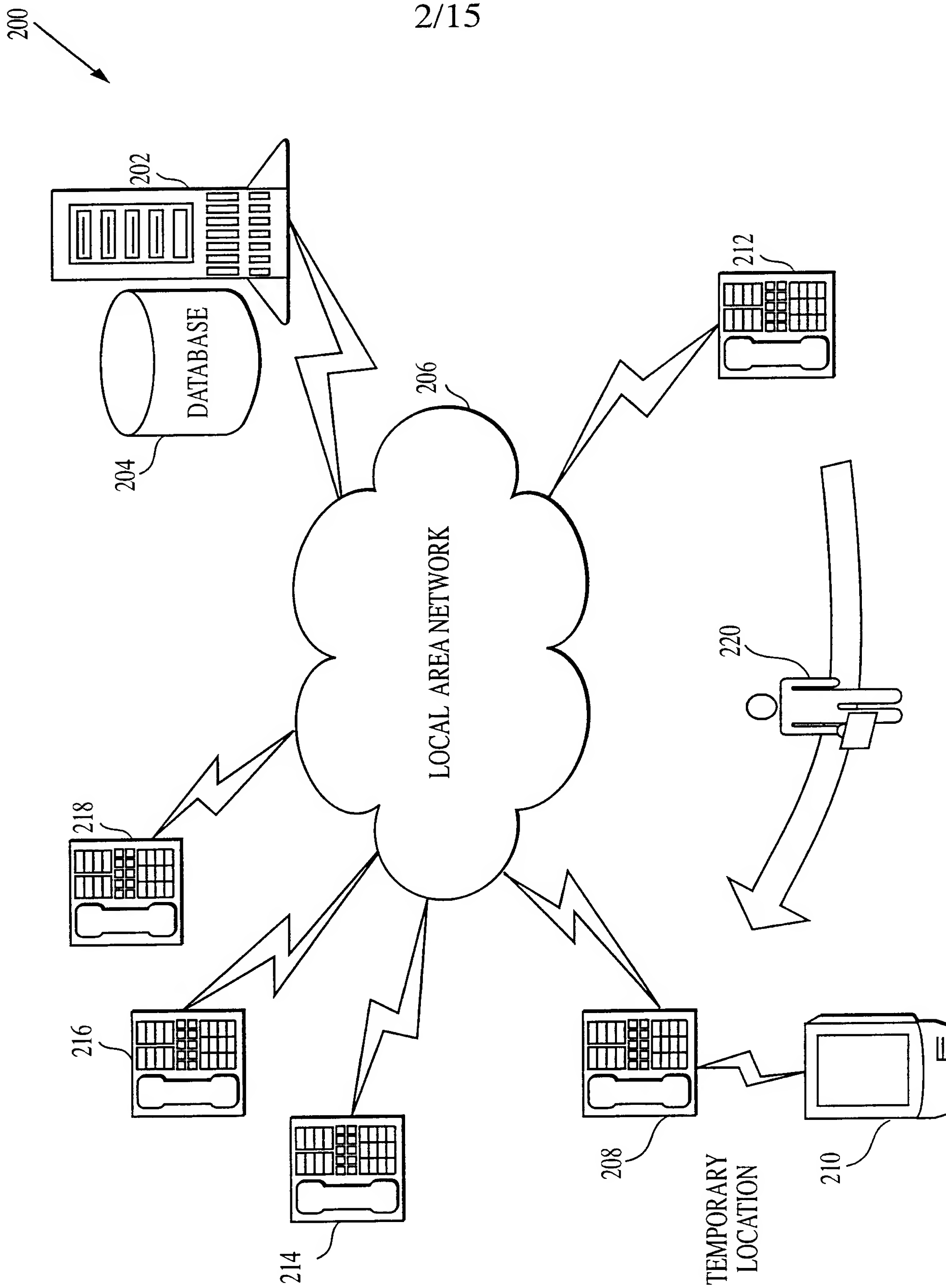
200
202
DATABASE
204
206
LOCAL AREA NETWORK
212
220
218
216
214
208
TEMPORARY LOCATION
PID
210

FIG. 2

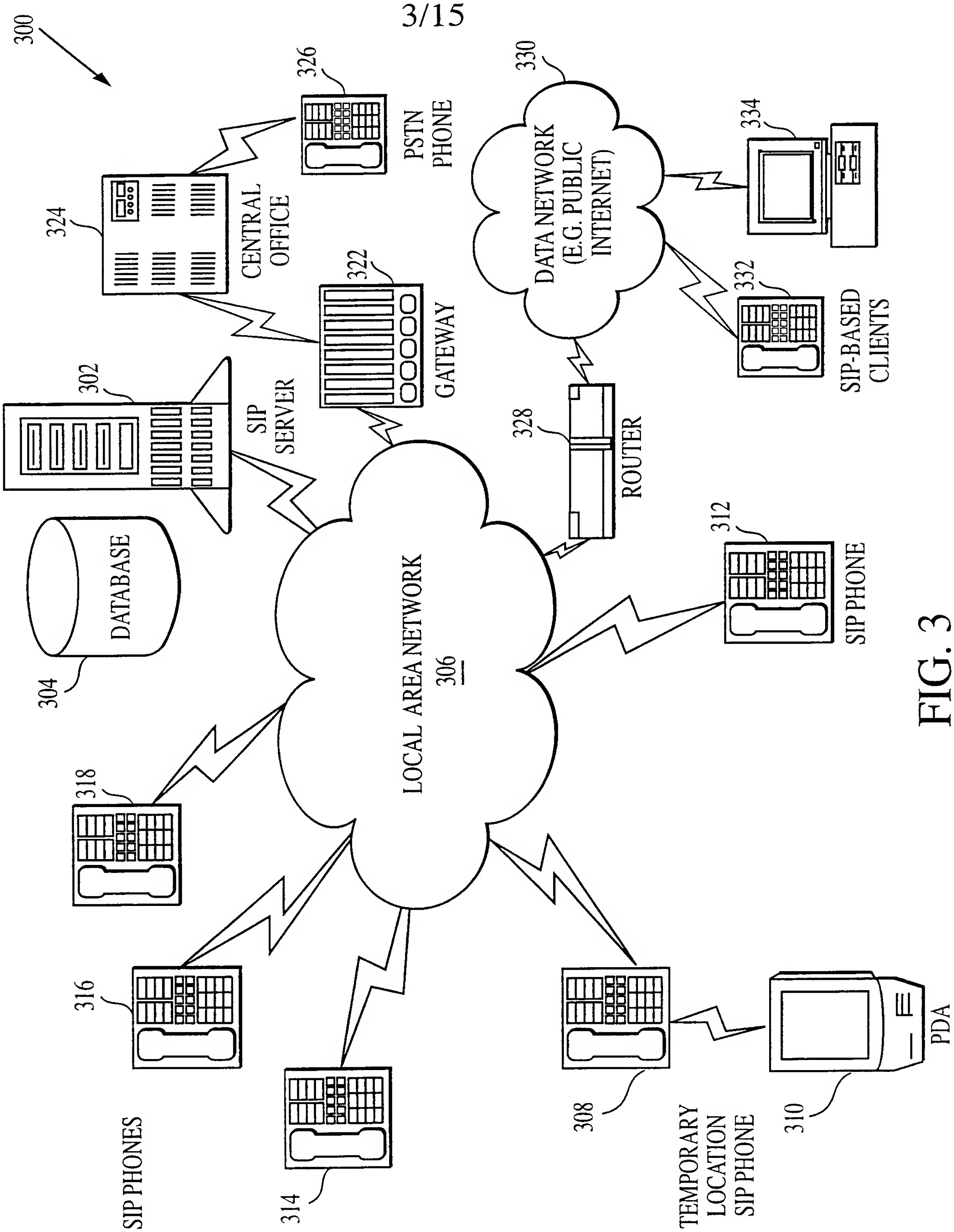
300
326
PSTN PHONE
330
DATA NETWORK (E.G. PUBLIC INTERNET)
334
324
CENTRAL OFFICE
322
GATEWAY
332
SIP-BASED CLIENTS
302
SIP SERVER
328
ROUTER
DATABASE
304
LOCAL AREA NETWORK
306
312
SIP PHONE
318
316
314
SIP PHONES
308
TEMPORARY LOCATION SIP PHONE
PDA
310

FIG. 3

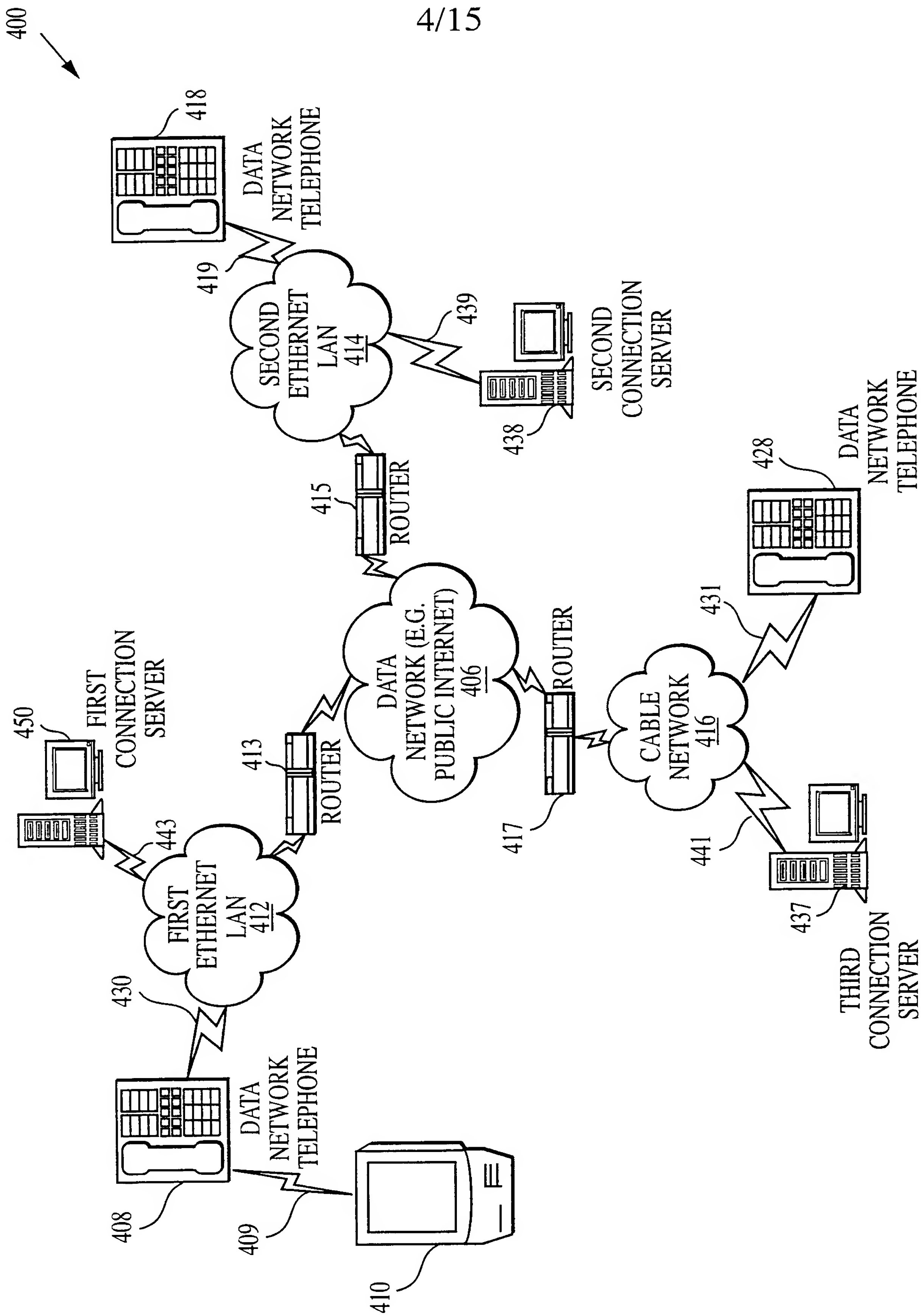
400
DATA NETWORK TELEPHONE
418
419
SECOND ETHERNET LAN 414
439
SECOND CONNECTION SERVER
438
415
ROUTER
DATA NETWORK TELEPHONE
428
FIRST CONNECTION SERVER
450
DATA NETWORK (E.G. PUBLIC INTERNET) 406
ROUTER
431
413
ROUTER
443
CABLE NETWORK 416
417
441
437
FIRST ETHERNET LAN 412
THIRD CONNECTION SERVER
430
DATA NETWORK TELEPHONE
408
409
410

FIG. 4

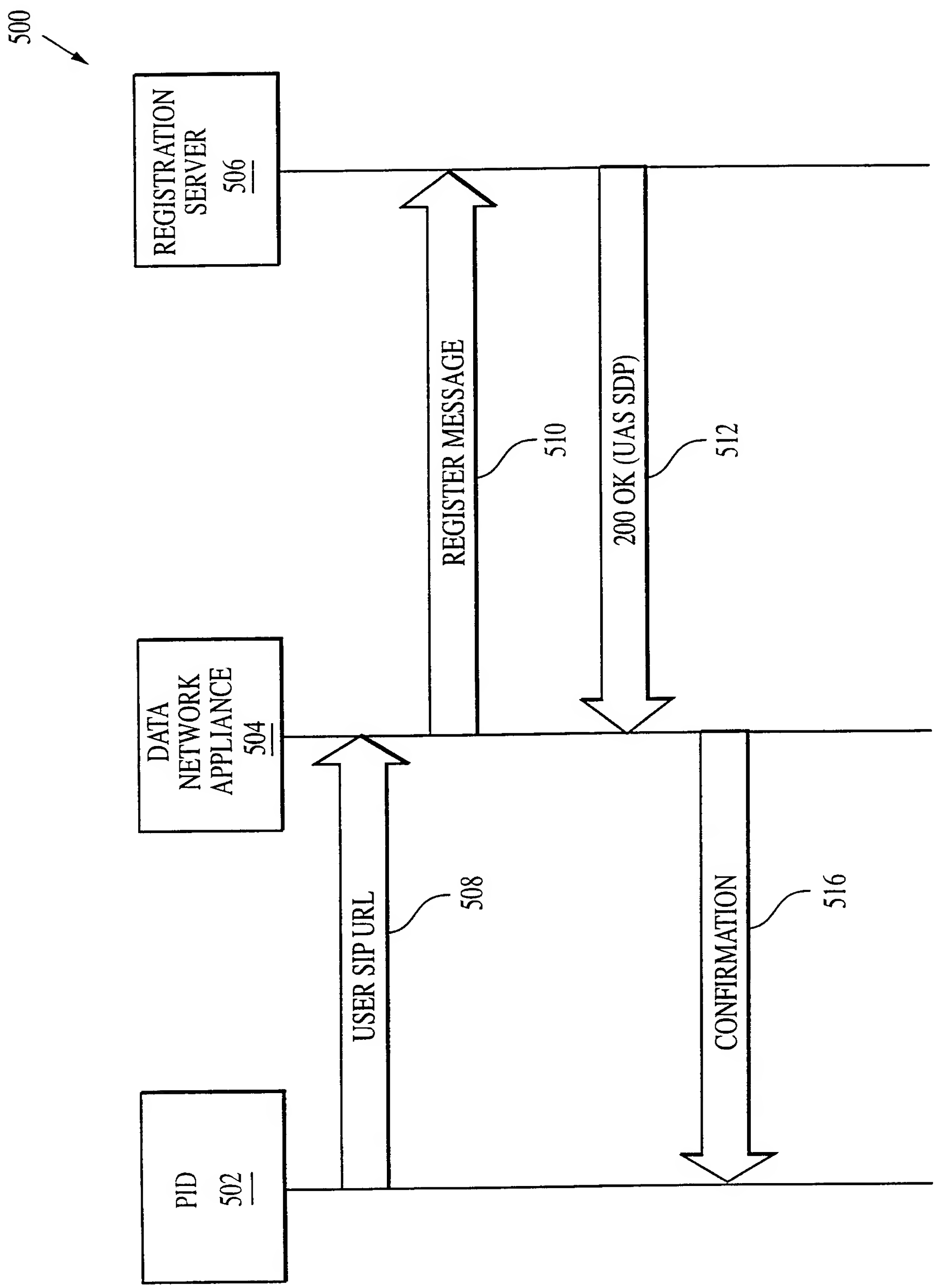
500
REGISTRATION SERVER 506
DATA NETWORK APPLIANCE 504
PID 502
USER SIP URL
508
REGISTER MESSAGE
510
200 OK (UAS SDP)
512
CONFIRMATION
516

FIG. 5A

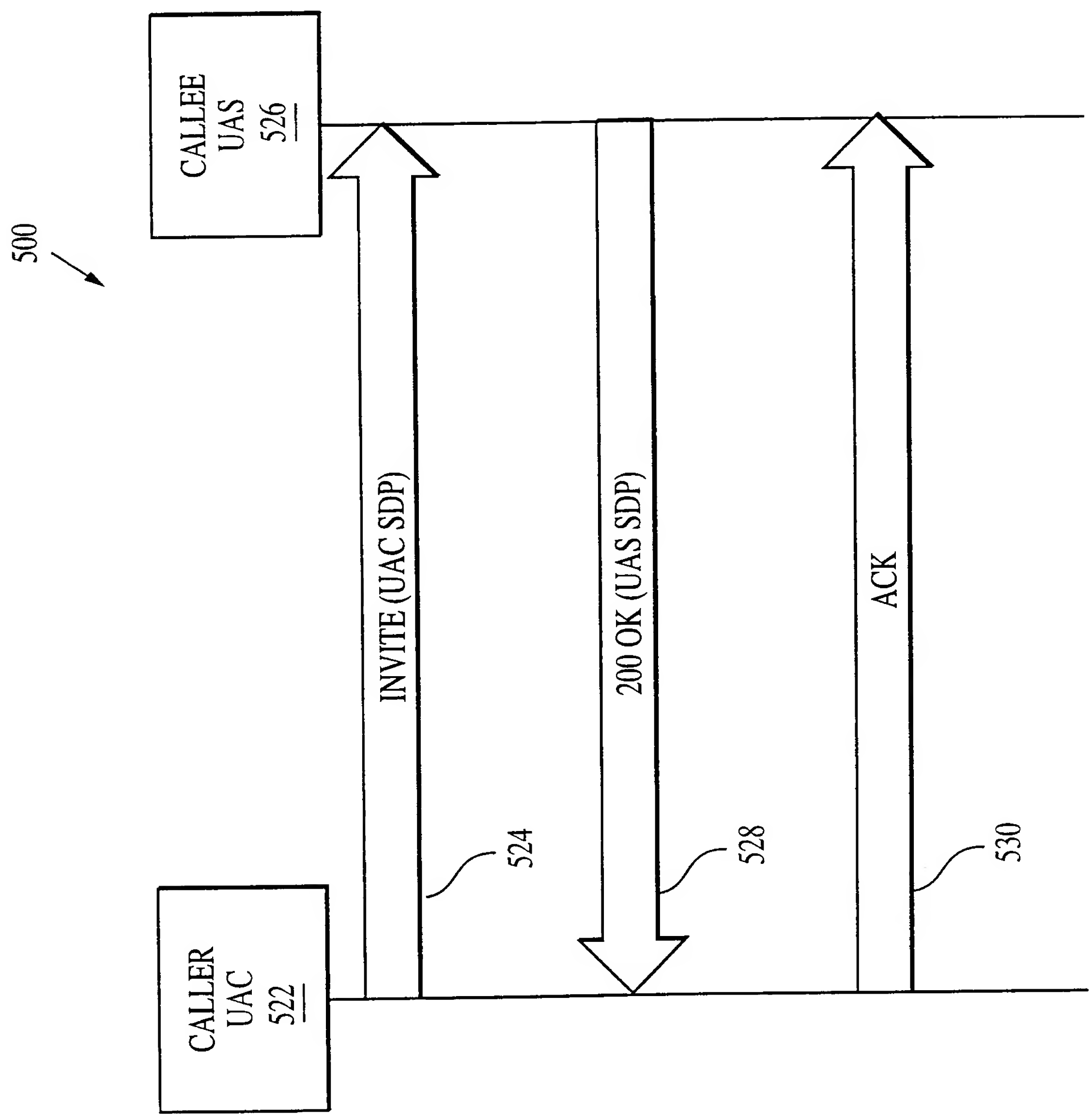
500
CALLEE
UAS
526
CALLER
UAC
522
INVITE (UAC SDP)
200 OK (UAS SDP)
ACK
524
528
530

FIG. 5B

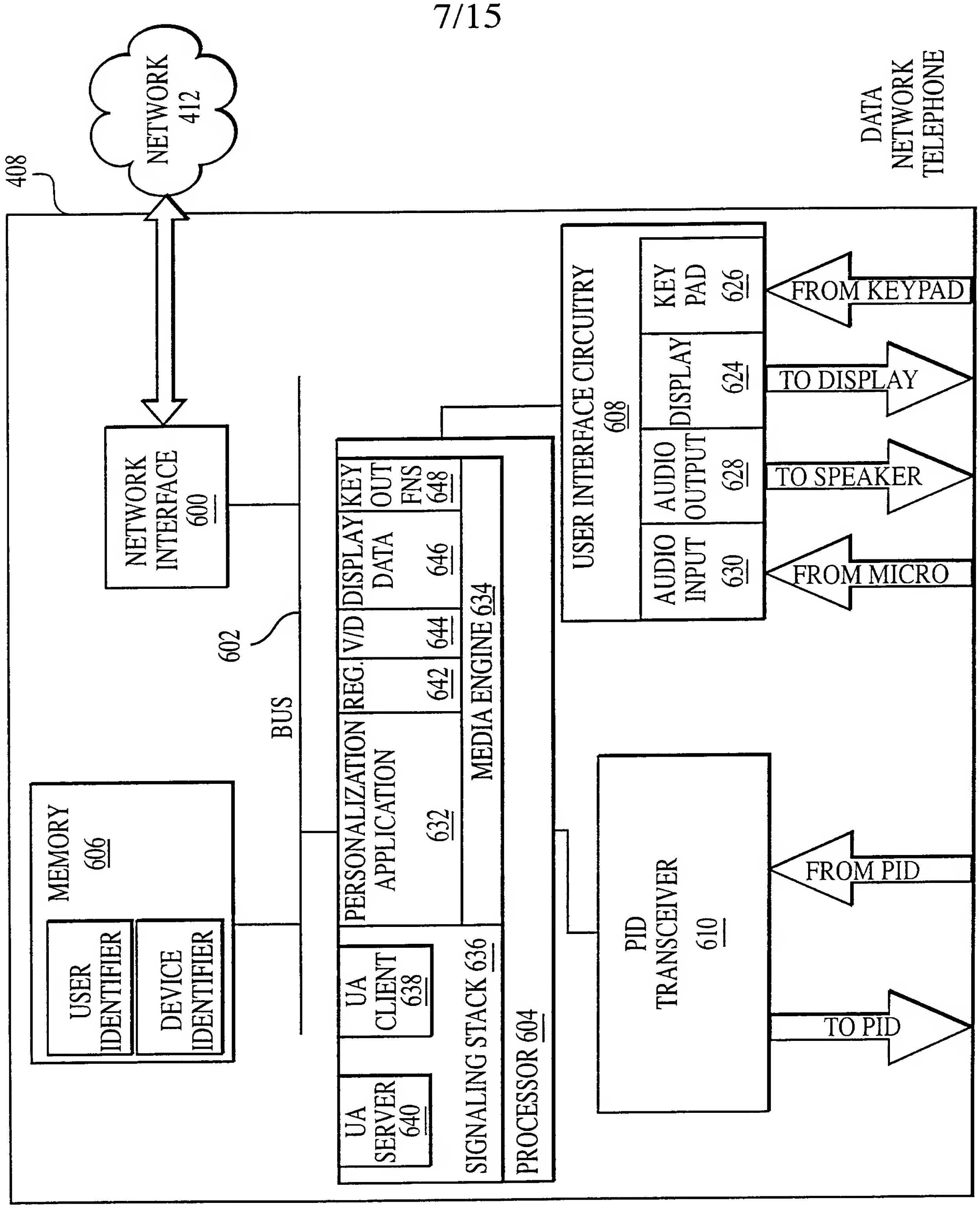
NETWORK
412
408
DATA
NETWORK
TELEPHONE
NETWORK
INTERFACE
600
BUS
602
MEMORY
606
USER
IDENTIFIER
DEVICE
IDENTIFIER
PERSONALIZATION
APPLICATION
632
REG.
642
V/D
644
DISPLAY
DATA
646
KEY
OUT
FNS
648
MEDIA ENGINE 634
UA
CLIENT
638
UA
SERVER
640
SIGNALING STACK 636
PROCESSOR 604
USER INTERFACE CIRCUITRY
608
KEY
PAD
626
DISPLAY
624
AUDIO
OUTPUT
628
AUDIO
INPUT
630
FROM KEYPAD
TO DISPLAY
TO SPEAKER
FROM MICRO
PID
TRANSCEIVER
610
FROM PID
TO PID

FIG. 6

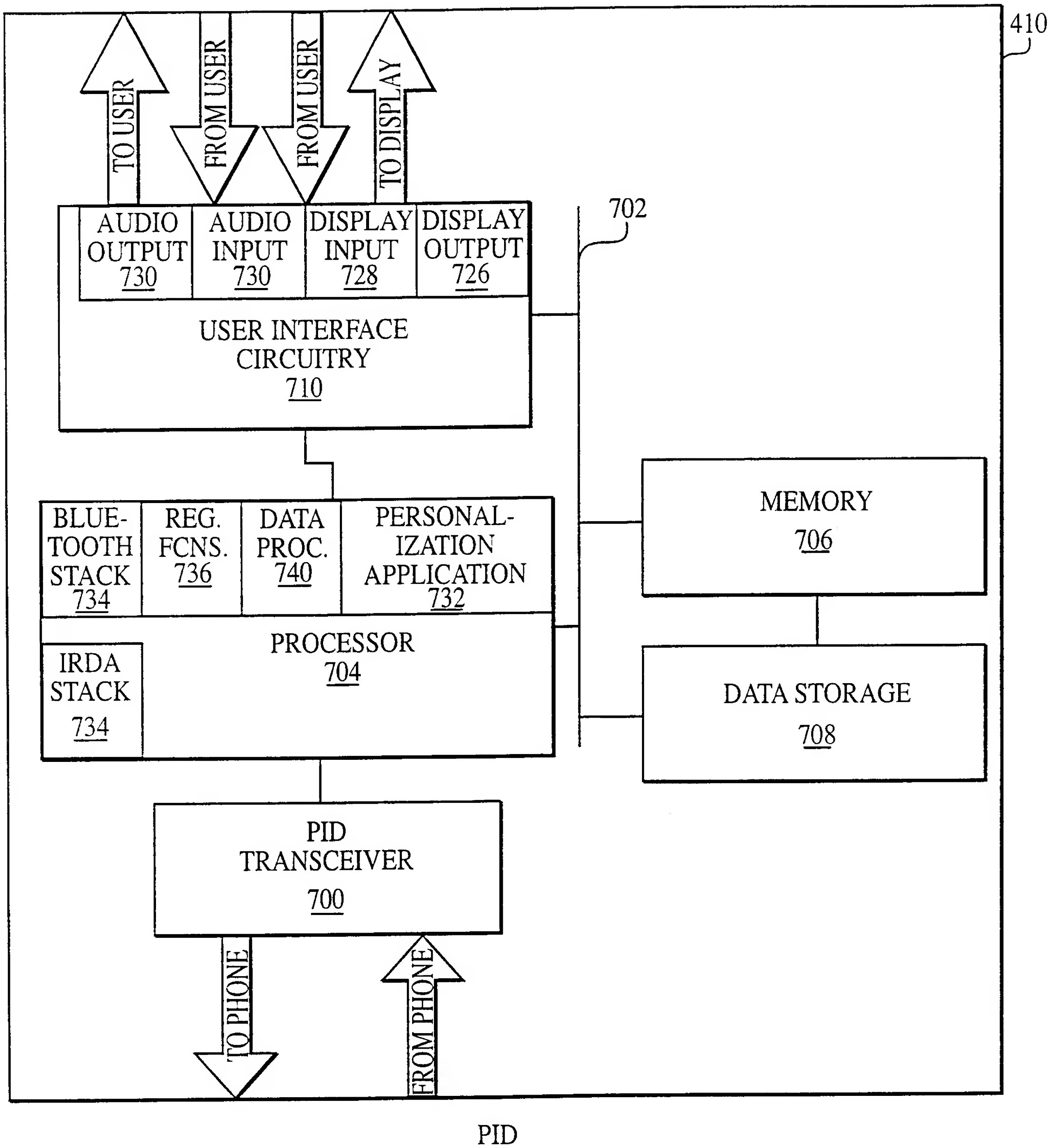
410
TO USER
FROM USER
FROM USER
TO DISPLAY
AUDIO OUTPUT 730
AUDIO INPUT 730
DISPLAY INPUT 728
DISPLAY OUTPUT 726
702
USER INTERFACE CIRCUITRY 710
BLUE-TOOTH STACK 734
REG. FCNS. 736
DATA PROC. 740
PERSONAL-IZATION APPLICATION 732
PROCESSOR 704
IRDA STACK 734
MEMORY 706
DATA STORAGE 708
PID TRANSCEIVER 700
TO PHONE
FROM PHONE
PID

FIG. 7

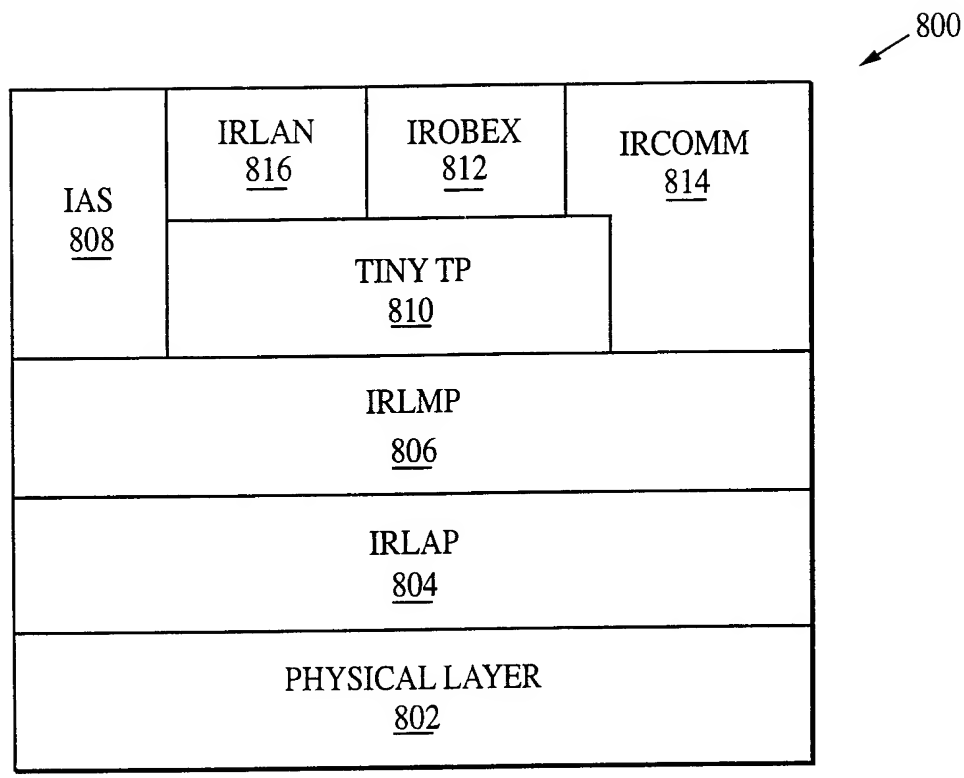
800
IAS
808
IRLAN
816
IROBEX
812
IRCOMM
814
TINY TP
810
IRLMP
806
IRLAP
804
PHYSICAL LAYER
802

FIG. 8

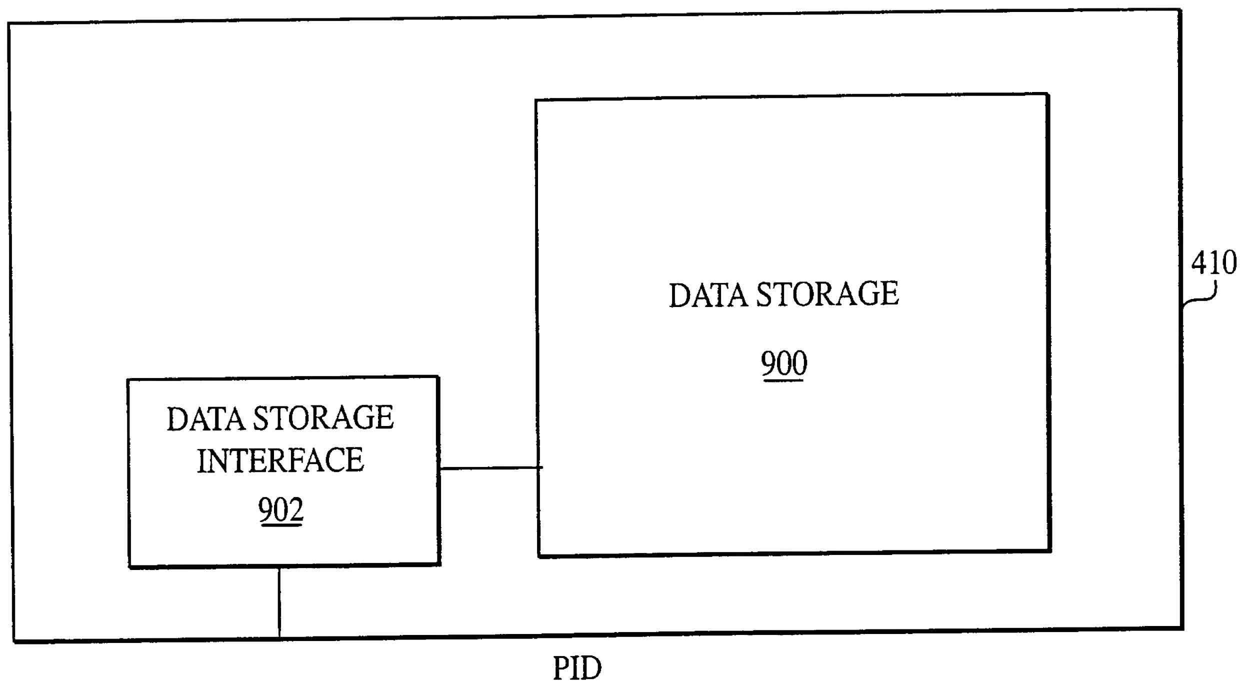

FIG. 9

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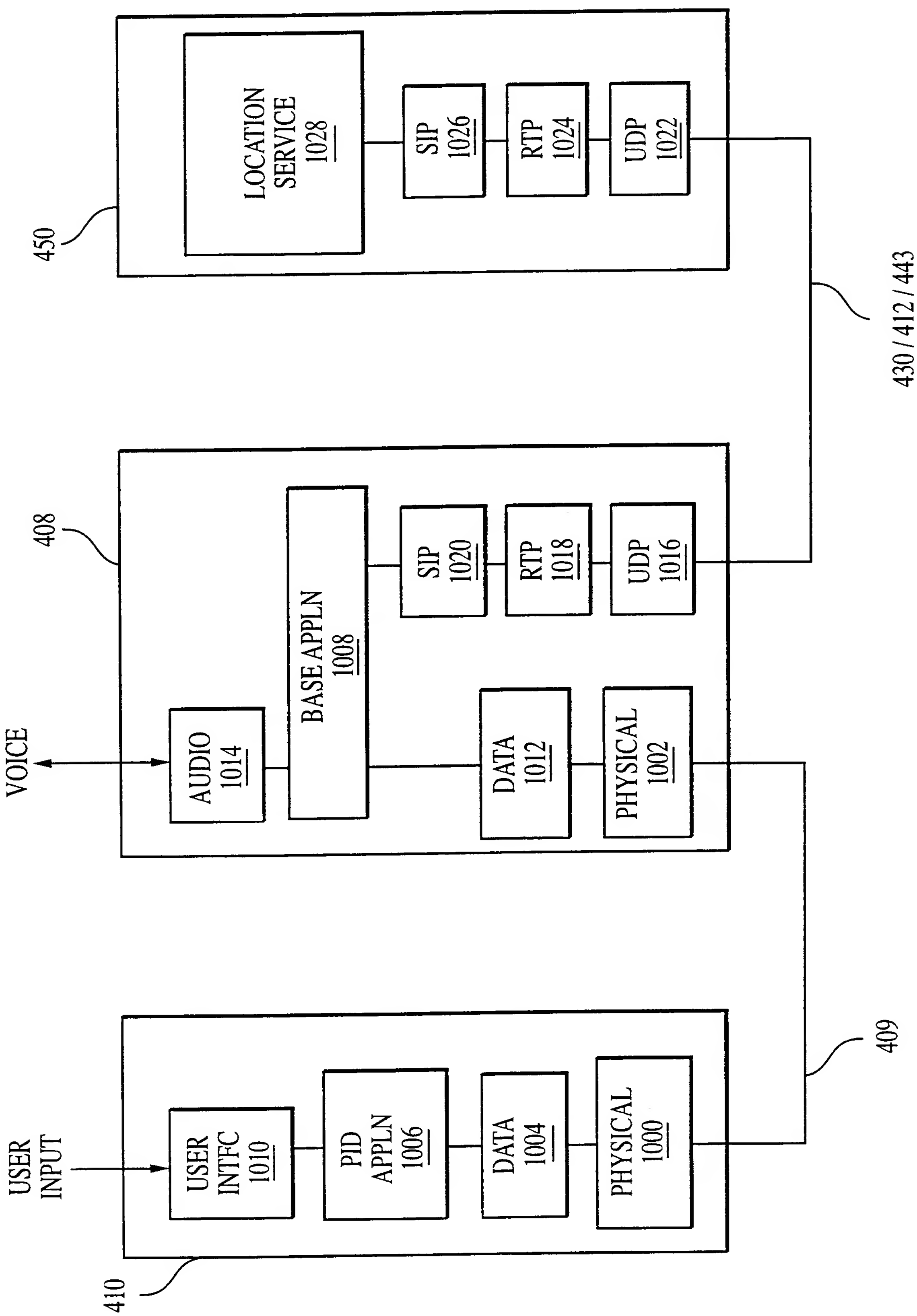
450
LOCATION SERVICE 1028
SIP 1026
RTP 1024
UDP 1022
430 / 412 / 443
408
VOICE
AUDIO 1014
BASE APPLN 1008
SIP 1020
RTP 1018
UDP 1016
DATA 1012
PHYSICAL 1002
409
USER INPUT
USER INTFC 1010
PID APPLN 1006
DATA 1004
PHYSICAL 1000
410

FIG. 10

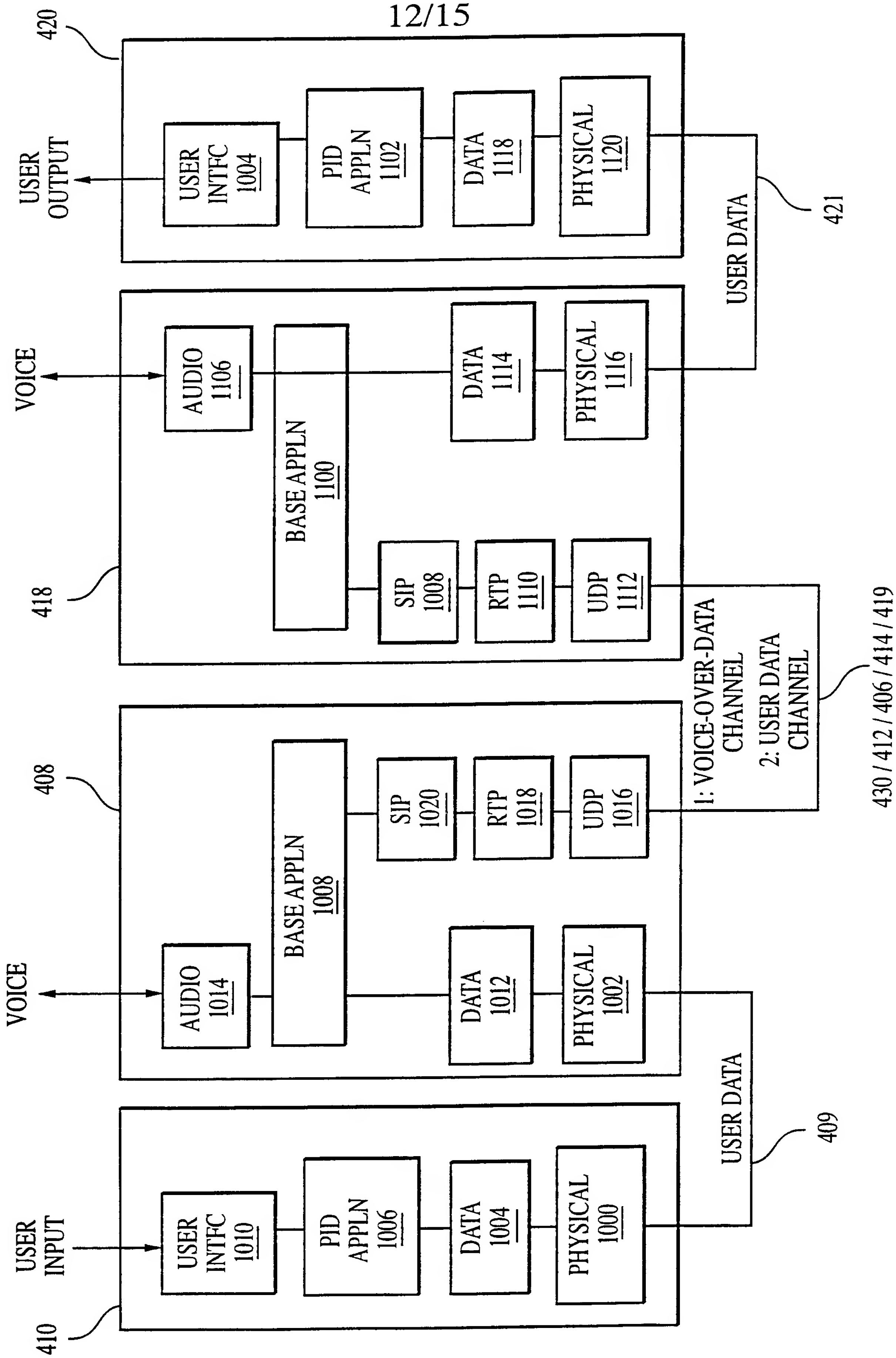
USER INPUT
USER INTFC 1010
PID APPLN 1006
DATA 1004
PHYSICAL 1000
410
USER DATA
409
VOICE
AUDIO 1014
BASE APPLN 1008
DATA 1012
PHYSICAL 1002
SIP 1020
RTP 1018
UDP 1016
408
1: VOICE-OVER-DATA CHANNEL
2: USER DATA CHANNEL
430 / 412 / 406 / 414 / 419
VOICE
AUDIO 1106
BASE APPLN 1100
DATA 1114
PHYSICAL 1116
SIP 1008
RTP 1110
UDP 1112
418
USER DATA
421
USER OUTPUT
USER INTFC 1004
PID APPLN 1102
DATA 1118
PHYSICAL 1120
420

FIG. 11

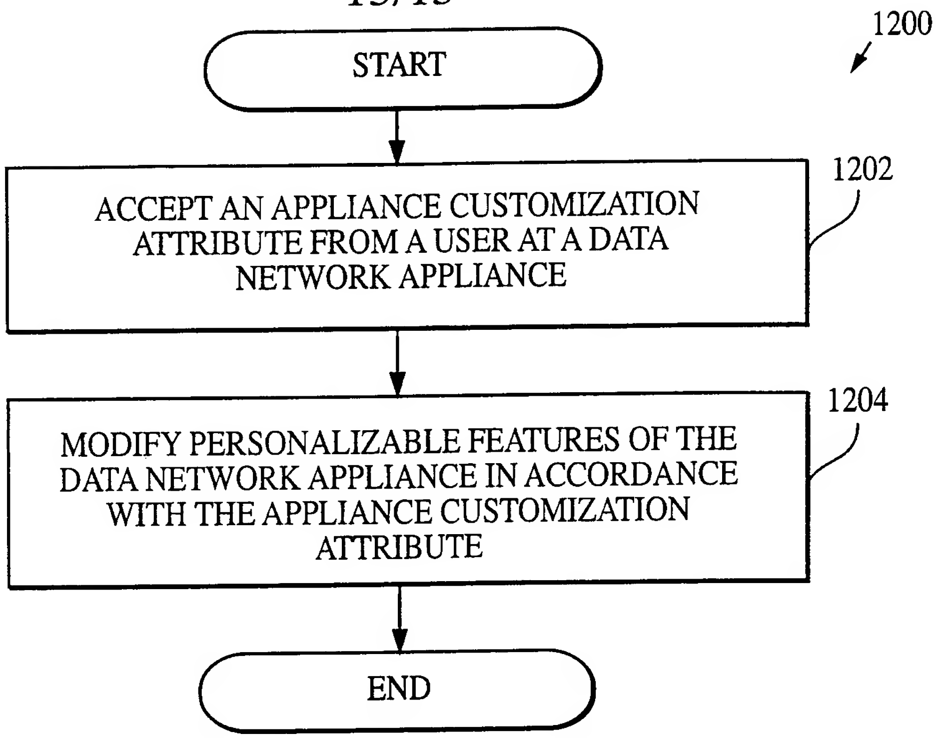
1200
START
ACCEPT AN APPLIANCE CUSTOMIZATION ATTRIBUTE FROM A USER AT A DATA NETWORK APPLIANCE
1202
MODIFY PERSONALIZABLE FEATURES OF THE DATA NETWORK APPLIANCE IN ACCORDANCE WITH THE APPLIANCE CUSTOMIZATION ATTRIBUTE
1204
END

FIG. 12

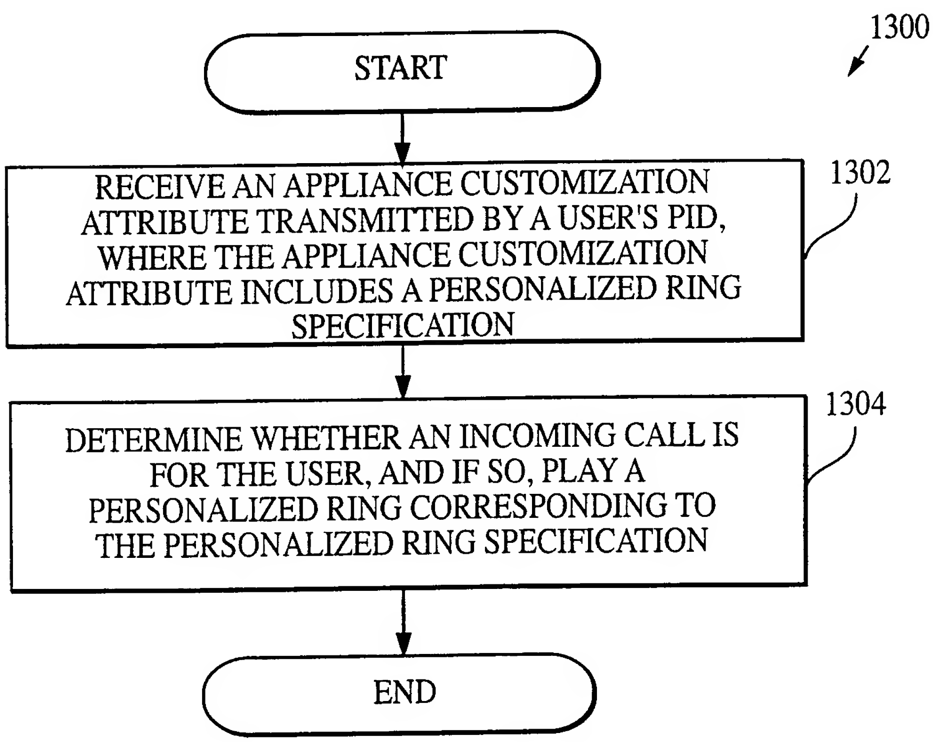
1300
START
RECEIVE AN APPLIANCE CUSTOMIZATION ATTRIBUTE TRANSMITTED BY A USER'S PID, WHERE THE APPLIANCE CUSTOMIZATION ATTRIBUTE INCLUDES A PERSONALIZED RING SPECIFICATION
1302
DETERMINE WHETHER AN INCOMING CALL IS FOR THE USER, AND IF SO, PLAY A PERSONALIZED RING CORRESPONDING TO THE PERSONALIZED RING SPECIFICATION
1304
END

FIG. 13

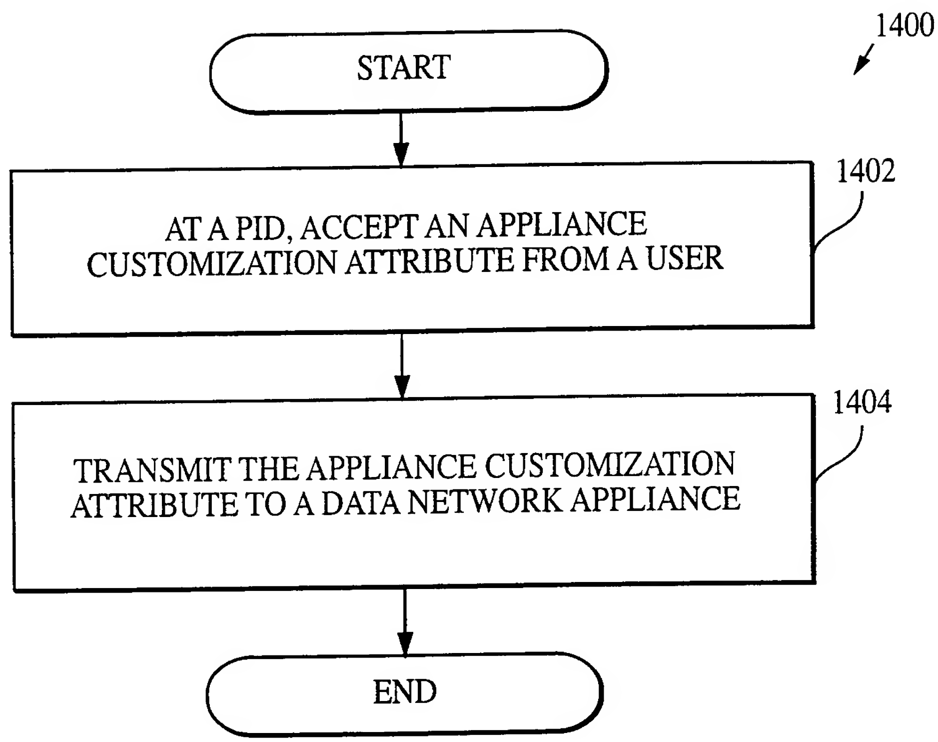
1400
START
AT A PID, ACCEPT AN APPLIANCE CUSTOMIZATION ATTRIBUTE FROM A USER
1402
TRANSMIT THE APPLIANCE CUSTOMIZATION ATTRIBUTE TO A DATA NETWORK APPLIANCE
1404
END

FIG. 14

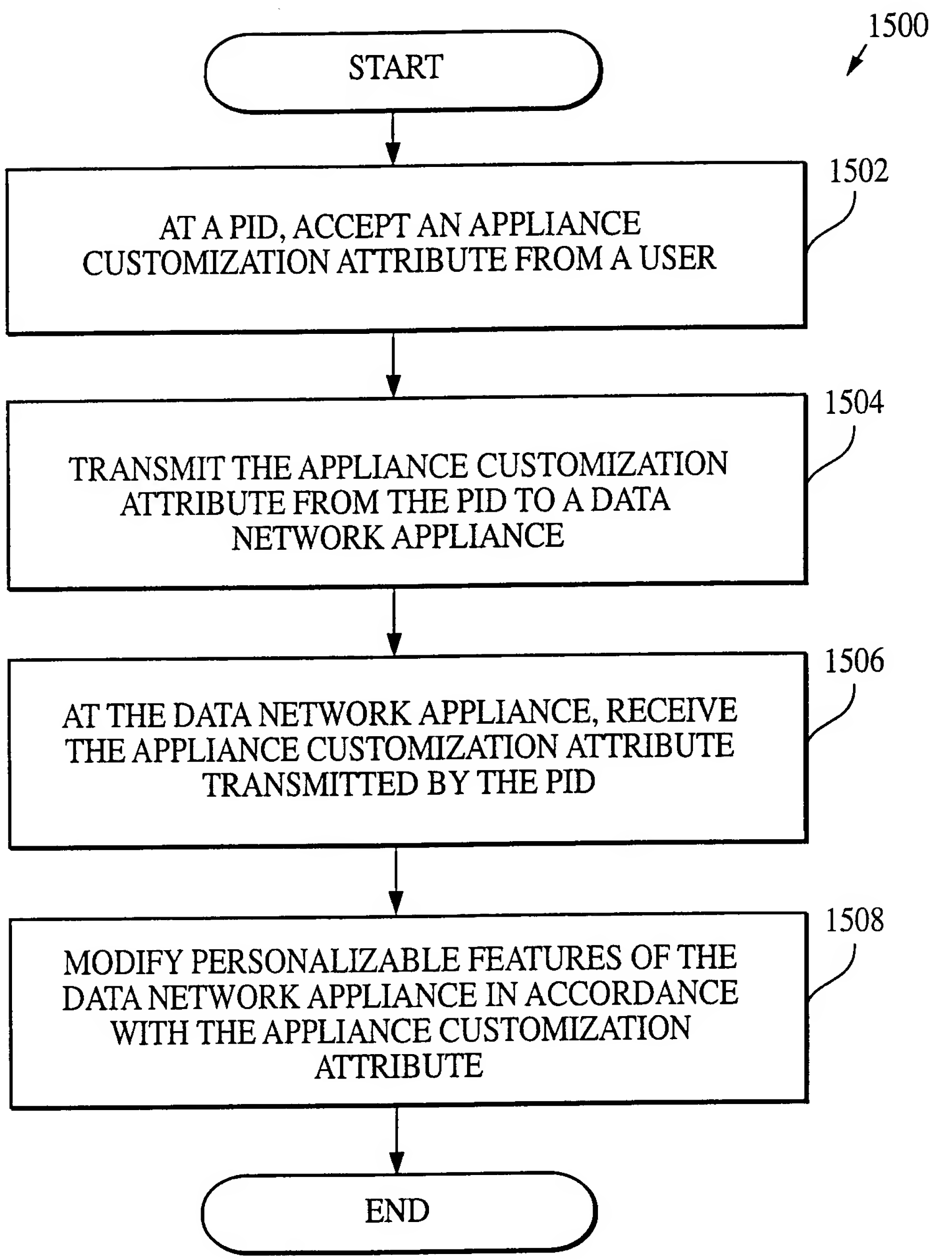
1500
START
1502
AT A PID, ACCEPT AN APPLIANCE CUSTOMIZATION ATTRIBUTE FROM A USER
1504
TRANSMIT THE APPLIANCE CUSTOMIZATION ATTRIBUTE FROM THE PID TO A DATA NETWORK APPLIANCE
1506
AT THE DATA NETWORK APPLIANCE, RECEIVE THE APPLIANCE CUSTOMIZATION ATTRIBUTE TRANSMITTED BY THE PID
1508
MODIFY PERSONALIZABLE FEATURES OF THE DATA NETWORK APPLIANCE IN ACCORDANCE WITH THE APPLIANCE CUSTOMIZATION ATTRIBUTE
END

FIG. 15